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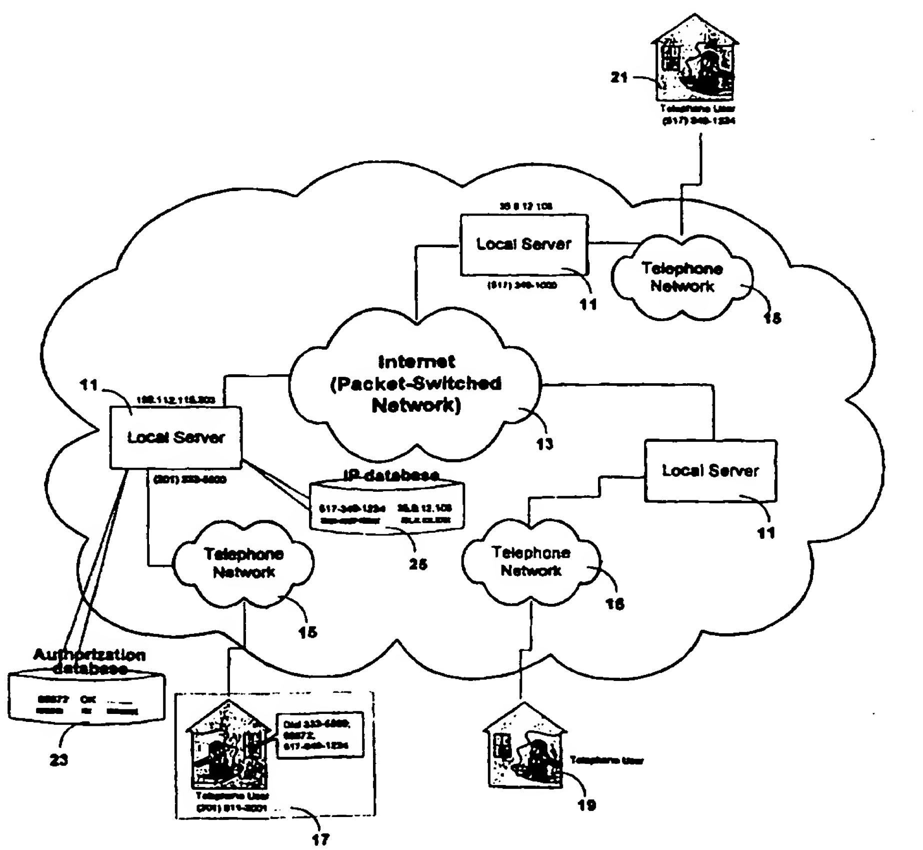
(54) Title: HYBRID NETWORK FOR REAL-TIME PHONE-TO-PHONE VOICE COMMUNICATIONS

(57) Abstract

A method and system are disclosed for permitting regular telephone users (1) to make long distance calls by way of a packet-switched digital data network (13) so as to avoid conventional long distance charges. Personal computers (PCs) are not needed, although they too may use the system. The system includes a plurality of geographically spaced servers (11), each associated with, for example, a particular area code. To make a long distance call, a user (17) simply dials the local number of the originating server (11), and optionally an authorization code (23). The user then inputs the telephone number of the recipient party to the originating server (11). The originating server (11) determines which remote server (11) in the system is local to the number being called and communicates with same via the digital data network (13). Thereafter, the addressed remote server (11) dials out the number of the recipient (19) and real-time voice communication is permitted between the caller (17) and the recipient (19). The system also includes other services such as group messaging, group fax, phone-to-PC communication, PC-to-phone communication, and PC-to-PC communication.



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HYBRID NETWORK FOR REAL-TIME PHONE-TO-PHONE VOICE COMMUNICATIONS

This invention relates to a system and corresponding method for permitting real-time telephone communication between parties via a packet-switched digital data network. More particularly, this invention relates to a hybrid communication network which utilizes an existing circuit-switched telephone network and an existing packet-switched network, the hybrid network including a plurality of geographically spaced servers interconnected via the packet-switched network enabling users to make "long distance" telephone calls by simply accessing their local server, which in turn automatically accesses another server local to the number being called and connects the calling and called parties.

BACKGROUND OF THE INVENTION

Figure 1 illustrates a conventional dedicated telephone network wherein "long distance" calls may be made from for example caller 1 to recipient 3 via network 5. Well known examples of such networks 5 are currently provided by AT&T™ and MCI™ as part of the Public Switched Telephone Network (PSTN). The switching technique of network 5 is based on circuit switching, i.e. each communication is afforded a "dedicated" channel for the duration of the communication. Because caller 1 and recipient 3 are located in different area codes, long distance charges are incurred by the caller upon long distance use of network 5. Unfortunately, these long distance charges quickly multiply and often become quite burdensome.

Long distance subscriber systems (e.g. see U.S. Pat. No. 4,513,175) competing with such established telephone company long distance systems have gained noteworthy acceptance. Typically, such subscriber systems employ the local switched telephone lines of an established telephone company to connect a subscriber to a

computer. The computer conveys the subscriber's telephone call over a dedicated transmission network to another local area where the call is again reintroduced into the switched telephone network and completed to the location dialed. The user is typically required to enter a seven digit local telephone number to gain access to the computer which controls the long distance dedicated network(s) to be employed. The computer answers the call and indicates that access has been gained by placing a tone or the like upon the line to the user. Upon hearing the tone, the user enters an assigned billing code, and thereafter, dials the area code and telephone number of the remote location which is to be contacted through the system.

Unfortunately, the long distance subscriber systems set forth in U.S. Patent No. 4,513,175 suffer from a number of problems. Firstly, the systems are not equipped to permit facsimile communication, multiparty conference calls, etc. as well as conventional telephone conversations. Secondly, these privately owned long distance networks are not packet-switched, and therefore suffer from the problems inherent in dedicated systems. Furthermore, the subscriber systems discussed in the '175 patent require the construction and maintenance of privately owned dedicated transmission media or lines. This is impractical and unduly expensive given today's marketplace.

U.S. Patent No. 5,341,374 discloses a token ring network integrating voice data and video with distributed call processing in a packet-switched network which supports real-time voice conversation. A plurality of token ring networks are interconnected via bridges or the like, each token ring network including a number of node coupling units (processor-controlled switches) arranged in a ring connected by a twisted pair. Each node may be coupled to a PC, telephone, and/or imaging system. Analog-to-digital (A/D) and digital-to-analog (D/A) conversion as well as data processing, display, and storage operations are

performed by the household devices (e.g. telephone, PC, etc.) coupled to the nodes. Unfortunately, the system of the '374 patent is not able to serve the majority of today's society because most households do not own PCs, facsimile machines, and digital telephones which perform A/D and D/A conversion in a MU-LAN PCM format. Households having just simple telephones which output analog voice signals during conversations cannot benefit from or utilize the '374 system. Also, phones not connected to the token-ring local area network (LAN) cannot use the system, i.e. the system is limited to token ring network technology which is undesirable given current market conditions.

U.S. Patent No. 4,969,184 discloses a data transmission system which utilizes a local public switched telephone network (PSTN) in transmitting information between remote data devices by way of a nationwide digital data network. A plurality of geographically spaced local nodes (each connected to a local PSTN) are connected via the digital data network enabling facsimile data, for example, to be transmitted from one area code to another via the digital data network without incurring long distance charges. Unfortunately, the system set forth in the '184 patent has numerous drawbacks, including (i) it is not capable of transmitting real-time continuous voice data; (ii) it requires the use of broadcast facilities and appears to be limited to facsimile or data transmission as opposed to voice transmission; and (iii) it requires the provision and maintenance of a private or paid-for digital data network.

In the last decade or so, the packet-switched digital data network commonly known as the "Internet" has gained popularity throughout the world. Figure 2 illustrates computer 7 communicating with computer 9 by way of the Internet 10. The Internet, the most known world-wide packet-switched network, is a collection of thousands of computer networks, tens of thousands of computers, and more than ten million users who share a

compatible means for interacting with one another to exchange digital data. The system is composed of many network providers interconnected via routers. The most commonly used method for transferring files is known as the file transfer protocol (ftp). Computers 7 and 9 typically access the Internet via various standard network interface cards, such as Ethernet and FDDI, or indirectly by way of data modems. Wire-type links are generally used.

The Internet is a packet-switched digital data network. Packet switching is a way in which different network segments can share a common transmission media. Rather than send a large block of data over a "dedicated" line directly to the destination computer, a packet switching network breaks the data into small chunks, each chunk being sent along a common transmission line in a "packet" that also contains source and destination information. This allows many packets to flow through the same network, all reaching their appropriate destination. Dedicated network components called packet-switching nodes route these packets from source to destination, using the information contained in the packet itself. After all the packets from a particular transmission of data reach their destination, the source and destination information is removed, and the packets are reassembled into the original data. In this way, packets from any number of computers can share the network.

Although it is currently unclear whether the following are prior art to the instant invention, systems which allow computer-to-computer voice communication over the Internet have recently been introduced into the marketplace. Using such systems, voice communication is possible over the Internet provided that the participating computers (PCs) are equipped with their own microphone, speaker, audio device, and necessary communication software. Unfortunately, this recently introduced computer-to-computer voice technology may only be utilized when both parties

have PCs equipped with the specialized hardware, software, and Internet connection. Furthermore, it is required that both parties be pre-notified of intended usage, and both computers be turned on with the necessary software before communication may take place. This is unduly burdensome and impractical as 70% to 80% of the households in the United States do not even have computers, not to mention the even higher percentage of non-computer households throughout the world.

International Discount Telecommunication (IDT) has recently announced a system for providing computer-to-phone voice communication over the Internet. Again, it is unclear at this time whether this system represents prior art to the instant invention. However, this computer-to-phone system also suffers from the problems set forth above regarding the computer-to-computer system and is further limited because it is not bi-directional. In other words, communications or voice conversations can only originate from the PC. Callers who simply own a conventional telephone (i.e. hook and ring device) may not call either PC owners or other phone owners by way of this system. This is a problem.

In view of the above, it is clear that there exists a need in the art for a bi-directional system and corresponding method for permitting real-time voice conversation between telephone users (without the need for PCs or the like) wherein any telephone owner or caller who desires to make a long distance call simply dials a local number which results in real-time voice communication between the caller and recipient via a digital packet-switched network thereby eliminating the incurrence of conventional long distance charges. There also exists a need in the art for such a system which will also support facsimile (fax) transmissions as well as multi-party or conference calls.

SUMMARY OF THE INVENTION

Generally speaking, this invention fulfills the above-described needs in the art by providing a bi-directional telecommunication network enabling real-time voice communication between callers and recipients, the telecommunication network comprising:

a plurality of bi-directional communication servers interconnected by way of a packet-switched digital data network, each of the servers being coupled to users by way of a switched telephone network so that a caller can access an originating server over the telephone network and input a destination telephone number of a recipient; and

wherein each of the servers includes means for receiving one of the destination telephone numbers from a caller and in response establishing real-time voice communication between the caller and the recipient via the destination server over the packet-switched digital data network.

According to certain preferred embodiments, the system also enables facsimile, group facsimile, multi-party voice, and PC-to-PC communication.

This invention further fulfills the above-described needs in the art by providing a method of making a long distance telephone call in real time from a caller to a recipient, the method comprising the steps of:

a) providing a first server local to the caller and a second server local to the recipient, the first and second servers being connected to one another by a digital data network;

b) the caller dialing a local telephone number in order to access the first server by way of a local switched telephone network;

c) the caller selecting a two party voice communication mode from a plurality of possible modes, the other possible modes including a facsimile mode and a PC-to-PC mode;

d) the caller entering the recipient's telephone number which is received by the first server;

e) upon receipt of the recipient's telephone number, the first server instructing the second server via the digital data network to call the recipient;

f) the second server calling the recipient's telephone number by way of a local call in order to connect the caller and recipient via the first and second servers and the digital data network; and

g) the caller and recipient carrying on a real-time voice telephone conversation during which the first and the second servers each perform D/A and A/D conversion of voice signals thereby enabling the parties to carry on the conversation using telephones which output analog voice signals.

In addition to phone-to-phone communication, the system also permits phone-to-PC, PC-to-phone, and PC-to-PC communications, provided that the PCs have an audio device, speaker, microphone, and software to implement same.

This invention will now be described with respect to certain embodiments thereof, accompanied by certain illustrations wherein:

IN THE DRAWINGS

Figure 1 is a prior art illustration of a conventional PSTN system permitting long distance telephone calls between a caller and recipient.

Figure 2 is a prior art illustration of a pair of computers communicating with one another via a packet-switched digital data network such as the Internet.

Figure 3 is a block diagram of a hybrid communication network utilizing existing telephone networks and an existing packet-switched digital data network according to this invention,

the hybrid network including a plurality of geographically diverse bi-directional servers interconnected by the packet-switched network.

Figure 4 is a block diagram illustrating a communication server of the Figure 3 system.

Figure 5 is a block diagram of the voice/data/fax controller of the Figure 4 server.

Figure 6 is a flowchart illustrating how a calling party or caller utilizes the Figures 3-5 system in order to choose between one of multiple different modes of communication.

Figure 7 is a flowchart of the two-party voice mode shown in Figure 6.

Figure 8 is a flowchart illustrating functionality and/or steps associated with the multi-party modes of Figure 6.

Figure 9 is a flowchart illustrating steps carried out by a calling or originating server (i.e. server local to the caller).

Figure 10 is a flowchart of steps carried out by an originating server in facsimile, group facsimile, and group messaging modes.

Figure 11 is a flowchart illustrating steps carried out by an originating server in the two-party voice mode.

Figure 12 is a flowchart illustrating steps carried out by an originating server in the multi-party conferencing mode.

Figure 13 is a flowchart illustrating the functions performed by the servers in the network in both the reception and transmission modes.

Figure 14 is a flowchart illustrating dialing out steps performed by a destination server local to the recipient.

Figure 15 is a flowchart illustrating dialing out functions performed by the called or destination server when real-time communication is not required between the caller and recipient.

DETAILED DESCRIPTION OF CERTAIN EMBODIMENTS OF THIS INVENTION

Referring now more particularly to the accompanying drawings in which like reference numerals indicate like parts throughout the several views.

Figure 3 illustrates a hybrid network for providing real-time telephone voice communication between remotely located callers and recipients, the hybrid network utilizing existing circuit-switched telephone network(s) 15 having dedicated lines and existing packet-switched digital data network 13 (e.g. the Internet). The hybrid network permits callers 17, 19, or 21 having simple telephones (and not a PC or facsimile machine) to make what would otherwise be long distance telephone calls to respective recipients without incurring the conventional long distance charge. The network uses no centralized control and combines the advantages of an existing local telephone network(s) 15 for cost effective local communication with the existence of, for example, the Internet 13 for economic global communication thereby allowing long distance telephone calls to be made without the usual "long distance" expense incurred when the PSTN is used. The hybrid network does not require callers and/or recipients of calls to have any "special" telecommunications equipment such as PCs, faxes, etc. other than a conventional analog-output telephone.

A caller accesses an originating server 11 using a local seven-digit telephone number and enters a recipient's number (destination telephone number including at least ten digits). The originating server looks up the destination number in its IP database 25 and determines the address of the corresponding server 11 local to the destination number (e.g. in the same area code). The originating server 11 then addresses and communicates with the destination server 11 over network 13,

which in turn calls the recipient over PSTN 15. When the recipient's telephone rings, the recipient simply picks up the phone and proceeds to conduct a regular phone conversation with the caller. In the case of voice messaging or multi-party conferencing, the recipient is notified of the type of service (or mode) by way of a voice message sent to the recipient from the destination server. In the case of a fax or group fax modes to be discussed below, the recipient is assumed to have a fax machine.

As shown in Figure 3, the hybrid network includes a plurality of geographically spaced communication servers 11 interconnected by way of packet-switched digital data network 13. According to certain embodiments, each server 11 is located in a different area code or local calling region. For example, Figure 3 illustrates the phone number of the server 11 local user 17 as (201) 333-5500 and the phone number of the server 11 local user 21 as (517) 349-1000. All servers 11 (e.g. PC-based including a Pentium™ processor) function as bi-directional interface devices between digital data network 13 and the switched telephone network 15 in that any one of households 17, 19, and 21 can communicate with one another no matter who originates the communication.

Figure 4 is a block diagram illustrating one of the plurality of servers 11 in detail. Each server 11 is connected to a corresponding local telephone network 15 by way of a private branch exchange (PBX) 16 so that a multiplicity of potential callers/recipients can access the system via each server. Alternatively, a channel service unit (CSU) may be used instead of PBX 16 to permit communication between network 15 and server 11. A standard T1 link 27 may be interposed between PBX 16 and server 11.

As shown, each household (17, 19, or 21) includes at least a standard analog-output telephone 29. Optionally, each household may also include a facsimile machine 31, personal computer (PC) 33, data modem 35, and/or wireless or cellular telephone 37. Each one of these devices may be used to access the hybrid network via an originating server 11 and the proximate local telephone network 15. If the user's phone 29 or PC 33 is equipped with a video display and/or camera, the system is able to support real-time audio/video conversation and imaging between callers and recipients.

Each server 11 includes buss or busses 39 which interconnects voice/data/fax controller(s) 41, storage 43, memory 45, processor(s) 47, and digital data network interface 49. Network interface 49 may be, for example, a conventional Ethernet or FDDI network access card. Multiple network adapter cards may be used when server 11 services many lines, the number of access cards required also being a function of the network bandwidth. Packetized data to be sent over network 13 may be formatted at 49 by way of conventional TCP/UDP/IP based protocols. For real-time voice communication, an efficient low-overhead UDP-based protocol is used. Optionally, the RTP (real-time transport protocol) or the public domain real-time audio transport protocol, vat, slightly modified, may be used.

Digital data storage 43 may include a standard storage disk while a Pentium-based chip(s) may be used in processor(s) 47. Storage 43 includes both authorization database 23 and IP database 25, as well as a directory database. Thus, information relating to which server 11 in the network (and its address) covers, or is local to, particular destination telephone numbers is stored at 43. For example, each server 11 in its storage 43 may include information indicating that if destination telephone number (517) 349-1234 is entered by a caller, then the network 13 address of the server 11 local to that particular number is

35.8.12.106 (see the telephone numbers and addresses shown in Figure 3). Additionally, storage 43 may be used to store accounting information, authorization code data, credit card information, and billing information relevant to particular users or households. Authorization database 23 maintains the authorization codes of active local users and their corresponding credit information. Meanwhile, memory 45 is utilized to store operating or application software used for controlling each server 11 by way of processor(s) 47. Additionally, data retrieved from storage 43 may be temporarily stored in memory 45 while calls and connections are being made.

The directory database within storage 43 maintains the personal directory of each user local to that server 11 (active and past users). For each user, the personal directory includes information such as the name and code of each group and individual which may be called in modes 85 and 87, personal usage information, personal billing information, transaction dates, etc. Because the directory database maintains records of both active and past users, when a past user wants to reactivate their account, the information is easily retrieved. According to certain embodiments of this invention, when a user moves from one area to another, the user's database information at 43 will be automatically transferred over network 13 from one server 11 to another server 11 local the new area, the transferring taking place either at the request of the user or when the user accesses his new originating server 11 for the first time.

By way of each user's personal directory database at the local server 11, the system according to this invention provides the following telephone services: (i) the user may check and delete voice messages left by others in his database; (ii) the user may check the status of group voice messages and faxes previously requested; (iii) directory information - the user may request a telephone number of a particular individual(s) if the

user inputs a name and location; (iv) the user may monitor his personal account, usage, etc.; and (v) the user can create, delete, and modify group names, codes and phone numbers relating to group and individual modes.

The duties or functions of processor(s) 47 include controlling the flow of data packets from controller 41 to network interface 49 and vice versa. Processor(s) 47 also controls the updating, retrieving, etc. of the billing data and the like stored at 43.

Voice/data/fax controller 41, provided in each server 11, is shown in more detail in Figure 5. Controller 41 includes fax/data modem 51, voice line interface 53, coder/decoder (CODEC) 55, digital signal processing unit (DSP) 57, DSP memory 59, compression/decompression device 61, encryption/decryption device 63, memory 65, and optionally processor(s) 67. The various devices shown in Figure 5 which make up each controller 41 are interconnected by way of buss 69 which communicates with buss 39.

Voice line interface 53 and fax/data modem 51 are connected to tone detector 52 which receives and properly distributes voice and/or fax/data signals which are incoming from PBX 16 over link 27. Accordingly, interface 53 receives from tone detector 52 incoming voice signals while modem 51 receives incoming fax/data signals. The detector 52 in controller 41 may be interfaced to the local switched (dedicated) telephone network 15 by way of a loop start (e.g. RJ 11 and/or RJ 14) when only a few voice lines are to be employed, while a standard T1 trunk 27 may be utilized for a larger number of lines (PBX 16 may be needed to distribute calls from the telephone network to an available line depending upon the number of lines being served). Each line can support both dial-in and dial-out functions (voice and/or fax) controlled by the voice processing board (see below).

CODEC 55 (e.g. Motorola MC145480 chip) performs standard analog-to-digital (A/D) and digital-to-analog (D/A) conversion. CODEC 55 functions to convert the analog signals received from interface 53 and/or modem 51 to digital signals (e.g. during a telephone conversation when the caller is outputting analog voice signals to the server via network 15).

On the other hand, because each server 11 is a bi-directional interface, when CODEC 55 receives digital data (e.g. digital voice data) from DSP 57, the CODEC converts it to analog, and thereafter forwards it to the local caller/recipient via either modem 51 or interface 53. Thus, CODEC 55 in each server 11 performs at least the following two functions: (i) converts analog signals incoming from its local caller/recipient to digital signals and forwards same over network 13 to the other party; and (ii) receives digital signals from the other party over network 13, converts the digital signals to analog signals, and forwards same to the local caller/recipient over the telephone network 15.

DSP 57 (e.g. TI TMS320 DSP family) performs sampling to voice grade frequency (e.g. 8 kHz) and may apply forward error correction (FEC) to the digital signals received from CODEC 55 in certain embodiments. DSP 57 performs digital echo cancellation and fax signal demodulation/modulation. DSP 57 also performs compression of the digital data to a lower number of bits (e.g. eight) per sample. In the other direction, DSP 57 functions to decode the error correction and decompress the digital data received from compression/decompression unit 61. DSP memory 59 stores information used in the error correction and compression/decompression processes performed by DSP 57.

Compression/decompression unit 61 performs a different type of compression/decompression than that done by DSP 57, thereby compressing data going out over network 13 and decompressing data coming from network 13. For example, unit 61 may utilize the

known GSM compression/decompression algorithm (about a 5 to 1 ratio). When security is of concern, encryption/decryption device 63 is provided and functions in a known manner (any standard encryption/decryption algorithm such as DES may be used) to encrypt digital data going out over network 13 and decrypt incoming digital data.

According to certain alternative embodiments of this invention, a Dialogic D/240SC-T1 24-port voice processing and T1 board may be utilized (this board including voice input, CODEC, DSP, DSP memory, and T1 connection) in conjunction with a Dialogic FAX/120 12-port fax board (including a fax modem and a fax data CODEC) to make up the above-listed components of controller 41. The Dialogic product supports half-duplex communication. A full duplex product, e.g. Calian VM200 high integration compressed voice/fax module, supports one port and performs the functions of steps 51, 52, 53, 55, 57, 59, and 61.

Processor(s) 67 is optional in that if provided, it controls the operation of the components shown in Figure 5 and the data flow therebetween. On the other hand, processor 67 is not required because processor(s) 47 (see Figure 4) may be utilized to perform these functions.

Beginning with Figure 6, certain embodiments of this invention will now be described by way of a call from a calling party (caller) to a receiving party (recipient) using the system of Figures 3-5. For the purpose of this description, let us assume that caller 17 (telephone number (201) 311-3001) wishes to telephone recipient 21 in a different area code at destination telephone number (517) 349-1234. In step 71, caller 17 begins the process by dialing the local telephone number (333-5500) of the proximate server 11 (originating server) so as to access the server by way of the local telephone network 15. At step 73, it is determined whether or not the local server number is busy. If so, the call is not made and the exit function 75 is performed.

However, if the connection between caller 17 and originating server 11 is made, the caller is prompted to enter an authorization code at 77. The caller may input the authorization code by way of DTMF signals or alternatively in a verbal manner. If the entered authorization code is verified, the caller is prompted to enter an input code at 79 for the purpose of selecting one of a plurality of possible different modes. If the authorization code is not verified, the exit function 75 is performed and the call terminated.

By entering the input code at 79, caller 17 may select one of the four different modes shown in Figure 6, namely, two-party DTMF input mode 81, two-party verbal input mode 83, multi-party DTMF input mode 85, and multi-party verbal input mode 87. The input code entered at 79 may be either verbal or DTMF when caller 17 is using a telephone.

When mode 81 is selected, the caller is prompted to enter a service code at 89 for the purpose of choosing one of the following four modes: i) miscellaneous personal services 91, such as personal directory information stored in the directory data base; ii) data modem mode 93 for PC-to-PC connection over network 13; iii) facsimile transmission mode 95; and iv) two-party real-time voice conversation mode 97. DTMF signals are used at 89 to select one of these modes when caller 17 is using telephone 29 or 37. In fax modes, DTMF inputs may be used at 79 and/or 89, while in PC-to-PC mode 93, the caller may prepend the authorization 77 and input 79 digits as a prefix to the telephone number of the originating server (these digits, once prepared, are saved in a file for automatic dialing).

When caller 17 wishes to utilize his PC in communicating with the recipient's PC, mode 93 is selected. Mode 95 is selected when the caller wishes to send a facsimile transmission to the recipient, while mode 97 is selected via DTMF when the caller wishes to engage in a real-time verbal phone conversation

with the recipient. When fax mode 95 is chosen at 89, caller 17 is prompted at 99 by the originating server 11 to enter the destination phone number of the recipient (e.g. (517) 349-1234), the use of this particular number assuming that the recipient's number is the same for both receiving fax and voice signals. Following step 99, the facsimile connection may be made and the fax sent at 101. Mode 93 also encompasses phone-to-PC and PC-to-phone communication in that caller 17 having a simple analog output telephone 29 may communicate in a real-time voice manner with a recipient having a PC equipped with audio receiving equipment, and vice versa, the PC having an address on packet-switched network 13. When, for example, caller 17 has a telephone and recipient 21 has such a PC, the caller dials the originating server 11 and at the same time inputs to the server (e.g. DTMF) the network 13 address of the recipient's PC. The originating server in turn communicates with the recipient's PC over network 13 thereby enabling real-time voice communication between caller 17 and the user of the PC. In a similar manner, caller 17 may utilize his PC 33 in calling recipient 21 having a simple telephone 29.

When two-party voice conversation mode 97 is chosen at 89, the caller is also prompted to enter the destination phone number (e.g. (517) 349-1234) of the recipient at 102. Thereafter, the destination server 11 local to the recipient is addressed by the originating server 11 via network 13 so that real-time two-party verbal communication may be made between the caller and recipient at 103.

When two-party verbal input mode 83 is chosen at 79, caller 17 is prompted to verbally input the destination phone number of the recipient at 105. Following step 105, the caller and recipient are connected as discussed above. Mode 83 may not be

utilized for facsimile purposes according to certain embodiments of this invention, but could be used in combination with PC mode 93.

When multi-party DTMF mode 85 is chosen at 79, the caller is prompted to enter a sequence of different destination phone numbers via DTMF, each complete telephone number being separated from the others by a "*" DTMF input, and the entire sequence ending with "**" at 107. In other words, the caller inputs a continuous sequence of destination telephone numbers (or codes), each number being separated from the adjacent number by a non-numeric DTMF input (e.g. "*" or "#"). Following the entering of the phone numbers of the parties to be called at 107, caller 17 is prompted to enter a service code at 109 for the purpose of selecting from among the three possible modes shown in Figure 6. Via DTMF, the caller may select from multi-party conferencing mode 111, group facsimile mode 113, and group voice message mode 115.

When multi-party conferencing mode 111 is selected at 109, caller 17 is connected by way of the required destination server(s) 11 to the multiplicity of recipients identified by the sequences entered in step 107 thereby resulting in a multi-party conference call. When mode 113 is selected at 109, the facsimile transmission entered by the caller is automatically sent to the plurality of destinations entered at 107 in a similar manner.

When group voice message mode 115 is selected at 109, a single voice message entered by caller 17 is transmitted to each destination telephone number or recipient identified at 107. In accordance with mode 115, caller 17 speaks the message to be sent at 117 and thereafter hangs up the phone at 119, the spoken message being recorded for later transmission by the originating server 11. After step 119, the originating server 11 determines from database 25 which other servers 11 in the hybrid network need to be contacted in order to communicate with each of the

telephone numbers entered at 107. After communication is made with each recipient, the voice message entered at 117 is sent to all recipients either simultaneously or at different times, depending upon the delay and/or traffic on network 13 (see below).

When multi-party verbal input mode 87 is chosen at 79, the caller is prompted to verbally input the group name and service type (voice message or conferencing) at 121. At 121, caller 17 may, for example, verbally enter the destination numbers of all recipients. Depending upon the input at 121, either mode 111 or 115 is chosen and carried out as discussed above.

It is important that the voice modes 103 and 111 be conducted between the caller and recipient(s) in substantially real-time. However, for voice messaging 115 and fax services 101, 113, which do not have stringent real-time requirements, a conventional file transfer protocol such as "ftp" may be used to transfer the message(s) to the destination server(s) at a time convenient to the servers and network. After the sending of a fax or voice message, the originating server receives at least one transmission status packet from the destination server(s) within a predetermined period of time (defined by the caller) indicating the status of the fax(es) or the voice message(s). In the case of fax services, the originating server faxes a status report to caller 17. For the case of voice messaging, the status is saved in storage 43 of the originating server in the form of a voice message so that caller 17 can check same at a later time via local switched telephone network 15.

Figure 7 is a flowchart illustrating possible responses to caller 17 using two-party voice mode 103 selected by way of either mode 81 or 83. As shown, following the initial communication between caller 17 and the destination server 11 via network 13, the caller waits for a response at 123. If a busy tone is heard 125, the caller simply hangs up the phone 127. On

the other hand, when the caller hears a ringing tone 129, a real-time verbal or voice conversation takes place at 131 between caller 17 and recipient 21 upon the recipient pickup up his/her phone (a message may be left on an answering machine if the recipient does not answer). Following conversation 131, each party simply hangs up the phone 127 and the exit function 129 is performed terminating the call. According to certain embodiments, simply the caller hanging up his phone will effect termination of the call.

Complications can arise while caller 17 is waiting for a response at 123. When it is determined by the originating server 11 that all network 13 lines are busy 133, a pre-recorded message is played to the caller indicating that the caller should switch to a regular telephone service, such as AT&T or MCI (PSTN). When originating server 11 determines that there is a bad communication via either network 13 or the remote telephone network 15, at 135, a similar pre-recorded message is played to the calling party advising a switch to conventional telephone service 137. Such a "bad communication" message could, for example, result from a caller-to-recipient network 13 delay which exceeds a predetermined threshold (see Fig. 11). Following the playing of such a message to caller 17 at 137 in response to one of findings 133 and 135, the caller may opt to have the originating server 11 automatically switch the caller to regular long distance service via PSTN 15. If the caller chooses this option, then the call is forwarded at 139 to the recipient's telephone number via the PSTN. If the caller in response to the message at 137 chooses not to be connected via conventional long distance service, then exit function 141 is performed and the call terminated.

Depending upon the number of servers 11 in the hybrid network located throughout the country or throughout the world, it may be the case that the telephone number of the recipient being dialed is not local to a particular server 11 (i.e. the destination number is not found in server database 25). If such is the case, it is determined by the originating server at 142 at which time a pre-recorded message is played to the caller at 137 asking whether or not the caller wishes to be switched to the PSTN as set forth above.

Figure 8 is a flowchart of multi-party conferencing mode 111 as selected by way of either mode 85 or 87. When mode 111 is selected, caller 17 waits for a response at 143. When it is determined by the originating server 11 (i.e. the server local to the calling party) at 145 that all parties identified in either step 107 or 121 are connected, the conference call is begun 147. After the conference call is over, the caller hangs up the phone 148 and the connection is terminated 149. However, when the originating server determines that one or a number of parties identified at 107 or 121 cannot be reached for one reason or another (e.g. line busy or excessive network delay), a voice message is played at 151 to the caller identifying which parties could not be connected. If all parties cannot be reached, caller 17 may simply terminate the call. Otherwise, the conference call may be started at 147 with only the parties which could be reached in attendance. Optionally, according to certain alternative embodiments of this invention, the parties which could not be connected at 151 may be accessed by the originating server 11 via a conventional long distance network (e.g. PSTN) and plugged into the conference call 147 with the parties accessed over the hybrid network.

Figure 9 is a flowchart illustrating the functionality of an originating server 11. As defined herein, an originating server is the server 11 local to and accessed by the calling party (caller). Upon connection between caller 17 and originating server 11, the server at 153 prompts the caller to input an authorization code. Upon receipt of the authorization code (e.g. DTMF), originating server 11 accesses at 155 its authorization database 23, 43, in order to determine if the authorization code is valid (whether it may be verified). When the server 11 determines at 155 that the authorization code input by the caller is improper or invalid, access to the hybrid network is denied at 157. However, if the server 11 determines that the authorization code input by the caller is valid, access to the hybrid network is authorized and originating server 11 prompts the caller at 159 to enter an input code in order to choose between the plurality of possible modes 81, 83, 85, and 87. Following step 159, the caller enters, for example, a DTMF input code (see reference numeral 79 in Figure 6) in order to select a mode of operation. As shown at 161, voice recognition and processing software is utilized when one of modes 83 and 87 is selected. Server 11 looks up in storage 43 (IP database 25) the remote server 11 address on network 13 covering or corresponding to the telephone number of the recipient (i.e. destination number). Select step 163 in Figure 9 encompasses the multiple steps shown in Figure 6 relating to mode selection. For example, steps 89, 107, 109, 121, etc. are included in service type identification step 163. Following step(s) 163, the different functions 91, 93, 101, 103, 111, 113, and 115 may be utilized as described above with respect to Figure 6.

Figure 10 is a flowchart illustrating the steps taken in fax mode 101, group fax mode 113, and group voice message mode 115 in the originating server 11. After one of modes 101, 113, and 115 is selected as shown in Figure 6, the originating server 11

receives the corresponding input from caller 17 by way of line 27 and saves it in either storage 43 or memory 45 at step 165. Thereafter, the dial-in line between caller 17 and server 11 is disconnected at 167. The originating server 11 takes the recipient's telephone number (e.g. (517) 349-1234) input from caller 17 and looks up in IP database 25 the appropriate server 11 which needs to be addressed. For example, as illustrated in Figure 3, the destination server 11 address corresponding to (517) 349-1234 is 35.8.12.106. This takes place at 169.

Following the determination by the originating server as to which server 11 needs to be addressed, the originating server sends file packets to each of the destination server(s) 11 at 171. Thereafter, the destination server(s) dials the recipient's number(s) input at 99, 107, or 121, and connects to the recipient. The originating server waits for a status update from the destination server(s) at 173. For example, when a single or group facsimile transmission is sent, the status is reported to the originating server at 175. Thereafter, caller 17 is free to dial the originating server 11 and determine the status of the fax (i.e. whether or not it was sent).

Figure 11 is a flowchart illustrating the steps taken by originating server 11 when two-party voice mode 103 is chosen by caller 17. Firstly, the server 11 receives and interprets the destination phone number (e.g. (517) 349-1234) entered by the caller at 177 and looks it up in its IP database at 179 to make sure that the hybrid system includes a server 11 local to that destination phone number. If IP database 25 lists a server address covering the received destination phone number (i.e. a match is found), then the originating server sends a connection request packet to the destination server 11 at 181. If the originating server at 179 determines that the hybrid system does not include a server 11 local to or covering the received destination phone number (i.e. no match is found), a voice

message is sent to caller 17 at 183 indicating that the destination phone number is not in the service area of the hybrid network. Thereafter, the call may be terminated 185.

After sending the connection request packet 181, the originating server 11 at 187 receives a reply packet from the destination server 11 indicating that either a connection has been made (or that all lines are busy). When all lines are found to be busy, the originating server sends an appropriate message to caller 17 at 189 and thereafter the call may be terminated 190.

When at 187 the originating server receives a reply packet from the destination server indicating that a connection has been made, the originating server at 191 compares the end-to-end network delay based upon the initial connection with a predetermined delay threshold in order to control the quality of real-time voice conversation. For example, if the predetermined threshold is 1.0 seconds, then it is determined at 191 by the originating server whether the end-to-end delay is greater than, or less than or equal to 1.0 seconds. If the delay is greater than 1.0 seconds (e.g. due to network congestion or the failure of the destination server), then a "bad communication" voice message is sent to caller 17 via telephone network 15 at 193. According to certain embodiments, the originating server 11 gives the caller the option (in the form of a voice message) to automatically dial the destination phone number through the regular PSTN following the "bad communication" message.

When it is determined at 191 that the end to end network delay between the originating server and the destination server is less than or equal to 1.0 seconds, then a full-duplex voice conversation takes place in real-time between caller 17 and recipient 21 at 195. Following the termination of the real-time

telephone call at 197, the length of the telephone call (the time of the call) is recorded in storage 43 so that caller 17 can be billed accordingly.

Figure 12 is a block diagram illustrating the steps taken by the originating server when multi-party conferencing mode 111 is selected by caller 17. At step 199, the server 11 makes a connection request to the requisite destination server(s) covering the destination telephone numbers entered at 107. An efficient multicast protocol such as the IP multicast protocol available on the Internet is used. Thereafter, in step 201, after the connection reply packets have been received, the originating server sends a voice message to caller 17 indicating which, if any, recipients or recipients could not be reached for the reasons discussed relative to Figure 7. At this time, the caller can either begin the multi-party conversation in real-time at 203 with the connected parties or hang up the phone which triggers the termination of all established connections. Following the termination of the multi-party conversation at 205, the originating server updates its billing records for caller 17. The caller is billed accordingly.

Figure 13 is a more detailed flowchart illustrating how a destination server handles a real-time full duplex voice conversation between caller 17 and recipient 21. The steps taken by server 11 transmitting signals over network 13 are illustrated on the left-hand side of Figure 13 while the steps carried out by server 11 in receiving signals over network 13 are illustrated on the right-hand side of Figure 13. In Figure 13, when a server 11 is the transmitting server, CODEC 55 digitizes received voice signals from the caller or recipient at 207. For example, CODEC 55 may utilize 8 KHz sampling and 8-bits per sample so that the controller generates 64K bits per second. Next, after CODEC 55 forwards the digital signal to DSP 57, compression device 61 compresses the digitized voice signal at 209 in order to reduce

network traffic (e.g. GSM compression algorithm). Thereafter, it is optional at 211 to utilize encryption device 63 to encrypt (e.g. DES) the compressed digital voice signal, depending upon whether security is of concern. From encrypter 63, the digitized signal is forwarded by way of buss 39 to network interface 49 where it is placed into a number of packets at 213 for transmission over digital data network 13. It is noted that compression/decompression and encryption/decryption may be performed either by special hardware chips (see Fig. 5) or by software executed by the processor(s) 47 in server 11. Multiple processors 47 may be needed if there are many lines to handle.

Thus, a server 11 acting in its transmitting mode sends the digitized packets at 215 through network 13 to the other server. At step 217, it is determined whether a hang-up signal has been sent (controller 41 is able to detect silent signals and hang-up signals). In the case of silent signals, no packet is sent so as to reduce network traffic. When a hang-up signal is detected, server 11 terminates the connection at 219 to the remote server 11, and thereafter updates the statistic information in storage 43 as to the connection time, called phone number(s), total number of packets transmitted, and the total number of packets dropped by the network.

The originating server 11 may continuously during a communication between a caller and recipient monitor the number of packets dropped or delayed over network 13 and compare the percentage to a predetermined tolerable threshold (e.g. 5%). If it is found that the percentage is greater than the 5% threshold, then a message is sent to the caller indicating that he will not be charged for the call.

Still with reference to Figure 13, we turn to the steps taken by a server 11 in the receiving mode. Firstly, the server receives packet data from network 13 at 221. Thereafter, server 11 assembles the packets at 223 and utilizes decrypting device 63

in order to decrypt the digital voice data at 225. Decompression device 61 then decompresses the digitized voice data at 227 and CODEC 55 converts the digital signal to analog at 229. When it is determined at 231 that a received packet from network 13 includes a hang-up signal, exit function 219 is performed.

Figure 14 is a flowchart illustrating the steps or functions performed by a called or destination server 11. Firstly, at 233, the server receives a connection request packet from an originating server 11 via network 13. The packet is interpreted in order to determine the type of request. When the request relates to a long distance call or the like (Figure 6), the destination phone number is extracted at 235. If the fax mode is selected, the server will try to allocate an available dial-out line 237, send the fax 239, and transmit a status packet back to the originating server at 241.

Meanwhile, when a voice messaging mode is selected, the dial out line is checked at 242. The received voice message is delivered over a dedicated line at 243 following the connection with an available line, and a status packet is sent back to the originating server at 244.

If a voice conversation mode is selected, it is determined at 245 whether dial-out lines are available. If all lines are busy, a message indicating same is sent back to the originating server at 246. If a phone line(s) is available and a connection is made with the destination phone number (e.g. (517) 349-1234), then the destination server at 247 sends a "connection established" packet back to the originating server. Thereafter, a real-time voice conversation takes place 248 and is terminated when desired 249.

For voice messaging and facsimile transmission modes, the real-time constraint is not stringent. Thus, if no dial-out line is available at 237 or 242, the destination server 11 will keep trying within a predetermined time period as shown in Figure 15,

which is a flowchart illustrating the steps performed in the dialing out to the recipient by the destination server 11. Firstly, the server searches for an available dial-out line at 251. When all are found to be in use, the destination server waits a predetermined period of time 252 before again searching for an available dial-out line. After the total waiting time breaks a predetermined threshold 253, the server sends a packet back to the originating server indicating that the connection could not be delivered after a predetermined period of time 254. When an available line is located at 251, the destination phone number (e.g. (517) 349-1234) is called 255. A determination is made at 256 whether the phone of recipient 21 is busy. If busy, the server proceeds to 252 while if answered, the connection is made between the caller and recipient and the routine is exited 257.

Once given the above disclosure, therefore, various other modifications, features, or improvements will become apparent to the skilled artisan. Such other features, modifications and improvements are thus considered a part of this invention, the scope of which is to be determined by the following claims.

WE CLAIM:

1. A method of making a real-time long distance telephone-to-telephone call from a caller to a recipient, the method comprising the steps of:

providing an originating communication server local to the caller;

providing a destination communication server local to the recipient;

interconnecting the originating server and the destination server via a packet-switched digital data network;

the caller telephoning the originating server using a local telephone number via a local switched telephone network, and thereafter communicating to the originating server a destination telephone number of the recipient;

the originating server forwarding to the destination server packetized digital data indicative of the telephone number of the recipient;

the destination server telephoning the recipient using a local telephone number via a switched telephone network and thereafter causing the caller and recipient to be connected for real-time voice conversation;

the originating server converting analog voice signals received from the caller to digital voice signals and thereafter forwarding same to the destination server in packet form via the digital data network during the real-time telephone conversation; and

the destination server receiving the digital voice signals from the originating server and converting same to analog voice signals and forwarding the analog voice signals to the recipient during the telephone conversation.

2. The method of claim 1, further comprising the steps of:

the destination server receiving analog voice signals from the recipient, converting them to digital signals and transmitting same to the originating server in packetized form during the conversation; and

the originating server receiving the packetized digital voice signals from the destination server, converting same to analog, and forwarding the analog voice signals to the caller via the telephone network.

3. The method of claim 1, further comprising the steps of:

determining a network delay between the originating and destination servers;

comparing the delay to a predetermined threshold delay time so that when the delay is less than or equal to the threshold, the conversation is permitted to take place.

4. A hybrid bi-directional telephone communication network for permitting real-time phone-to-phone long distance voice conversation between a first party and a second party, the hybrid network utilizing a circuit-switched telephone network and a packet-switched digital data network, the hybrid network comprising:

a first communication server local to the first party and coupled thereto via a switched telephone network, and a second communication server local to the second party and coupled thereto via a switched telephone network;

a packet-switched digital data network interconnecting and allowing packetized digital data communication between said first and second servers;

said first server including transmission mode means for: (i) receiving a local telephone call from the first party by way of the switched telephone network; (ii) receiving from the first party the telephone number of the second party; (iii) communicating with said second server over said digital data network and instructing said second server to call the telephone number of the second party; and (iv) converting analog voice signals received from the first party to digital signals and forwarding same to said second server in packetized form thereby enabling real-time telephone-to-telephone voice conversation between the first and second parties via the digital data network;

said first server further including receiving mode means for: (v) calling the first party upon receiving instructions to do so from the second server; and (vi) receiving digital voice signals from the second server via the digital data network and converting same to analog voice signals and forwarding the analog voice signals to the first party during the voice conversation;

said second server including transmission mode means for: (i) receiving a local telephone call from the second party by way of the switched telephone network; (ii) receiving from the second party the telephone number of the first party; (iii) communicating with said first server over the digital data network by way of data packets instructing the first server to call the telephone number of the first party; (iv) converting analog voice signals received from the second party to digital voice signals and forwarding same to said first server over the digital data network;

said second server further comprising receiving mode means for: (v) locally calling the second party upon receiving instructions to do so from said first server; and (vi) receiving packetized digital voice signals from said first server over said

digital data network and converting same to analog voice signals and forwarding the analog voice signals to the second party during the voice conversation; and

wherein said hybrid network including said first and second servers is bi-directional in that both the first and second parties are capable of initiating long distance telephone calls to the other using their respective telephones which output analog voice signals.

5. The hybrid network of claim 4, wherein each of said first and second servers further comprises facsimile mode means for allowing the first and second parties to send facsimile transmissions to one another over said digital data network.

6. The hybrid network of claim 5, wherein each of said first and second servers further comprises conference call means and group message means for allowing the first and second parties to conduct conference calls and send group messages respectively over the hybrid network via the digital data network; and

group facsimile means.

7. A method of making a long distance telephone call in real-time from a caller to a recipient, the method comprising the steps of:

a) providing a first server local to the caller and a second server local to the recipient, the first and second servers being connected to one another by a digital data network;

b) the caller dialing a local telephone number in order to access the first server by way of a local switched telephone network;

c) the caller selecting a two-party voice communication mode from a plurality of possible modes, the other possible modes including a facsimile mode and a PC-to-PC mode;

d) the caller entering the recipient's telephone number which is received by the first server;

e) upon receipt of the recipient's telephone number, the first server instructing the second server via the digital data network to call the recipient;

f) the second server calling the recipient's telephone number by way of a local call in order to connect the caller and recipient via the first and second servers and the digital data network; and

g) the caller and recipient carrying on a real-time voice telephone conversation during which the first and second servers each perform D/A and A/D conversation of voice signals thereby enabling the parties to carry on the conversation using telephones which output analog voice signals.

8. The method of claim 7, further comprising the step of determining a delay over the digital data network between the first and second servers, and comparing the delay with a predetermined threshold.

9. A bi-directional telecommunication network enabling real-time voice communication between callers and recipients, the telecommunication network comprising:

a plurality of bi-directional communication servers interconnected by way of a packet-switched digital data network, each of said servers being coupled to users by way of a switched telephone network so that a caller can access a local said server over the telephone network and input a destination telephone number of a recipient; and

wherein each of said servers includes means for receiving one of said destination telephone numbers from a caller and in response establishing real-time voice communication between the caller and the recipient via another said server over said packet-switched digital data network.

10. The network of claim 9, wherein each of said servers further comprises digital-to-analog conversion means for receiving analog voice signals from a local caller or recipient, converting same to digital signals, and thereafter transmitting said digital signals in packetized form over the digital data network to the other of said caller and recipient by way of another said server.

11. The network of claim 10, wherein each of said servers further comprises facsimile means for enabling callers to transmit facsimile data to recipients whereby facsimile transmissions originate from the caller, are forwarded to an originating said server over the switched telephone network, are thereafter packetized and sent to a destination said server over said digital data network, and forwarded to the recipient over the switched telephone network from said destination server.

12. The network of claim 11, wherein each of said servers further comprises group message means for enabling a voice message to be sent from a caller to a plurality of recipients, and group facsimile means for enabling a facsimile transmission to be sent from a caller to a plurality of recipients over the switched telephone network and the digital data network.

13. The network of claim 12, wherein each of said servers further comprises multi-party conferencing means for permitting a caller to initiate a conference call with a plurality of recipients over the switched telephone network and the digital data network.

14. The network of claim 9, wherein each of said servers further comprises delay determination means for determining the overall time delay from caller to recipient due to the digital data network, and comparison means for comparing the determined overall delay with a predetermined threshold delay time so that the call may be terminated when the overall delay exceeds the predetermined threshold delay time.

15. A method of making a telephone call comprising the steps of:

a caller telephoning a local server over a circuit-switched telephone network, and inputting the telephone number of a recipient;

the local server addressing a remote server over a packet-switched digital data network;

the remote server calling the recipient at the telephone number so that real-time phone-to-phone voice conversation is realized between the caller and recipient.

16. A method of a caller using a telephone calling a recipient PC, the PC being equipped with audio receiving equipment and having an address on a packet-switched network, the method comprising the steps of:

the caller dialing a local telephone number to access a server connected to the packet-switched network;

the caller inputting to the server the address of the recipient PC; and

the server addressing the PC over the packet-switched network thereby enabling real-time voice communication between the caller and a user of the PC.

17. The method of claim 16, wherein the caller enters the following sequences in the recited order to the server:

a) the local server telephone number;

b) an authorization code; and

c) the network address of the PC.

18. A method of making a conference call to a plurality of recipients comprising the steps of:

a caller accessing a server;

the caller inputting via DTMF a continuous sequence of destination telephone numbers corresponding to the recipients, each number being separated from the adjacent number(s) by a non-numeric DTMF input; and

the server receiving the continuous sequence of telephone numbers, and causing same to be dialed thereby permitting voice or fax communication between the caller and the plurality of recipients.

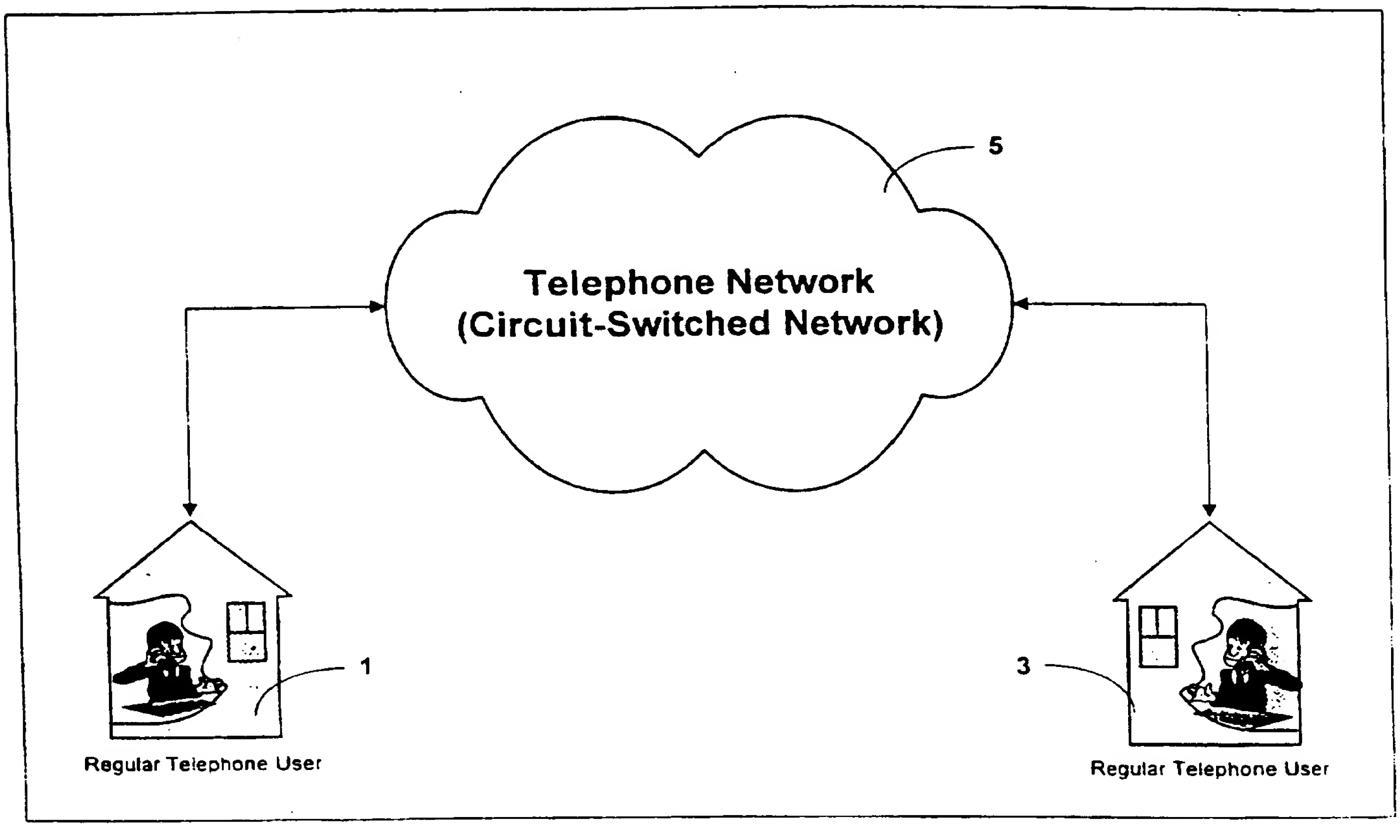


FIG. 1 (PRIOR ART)

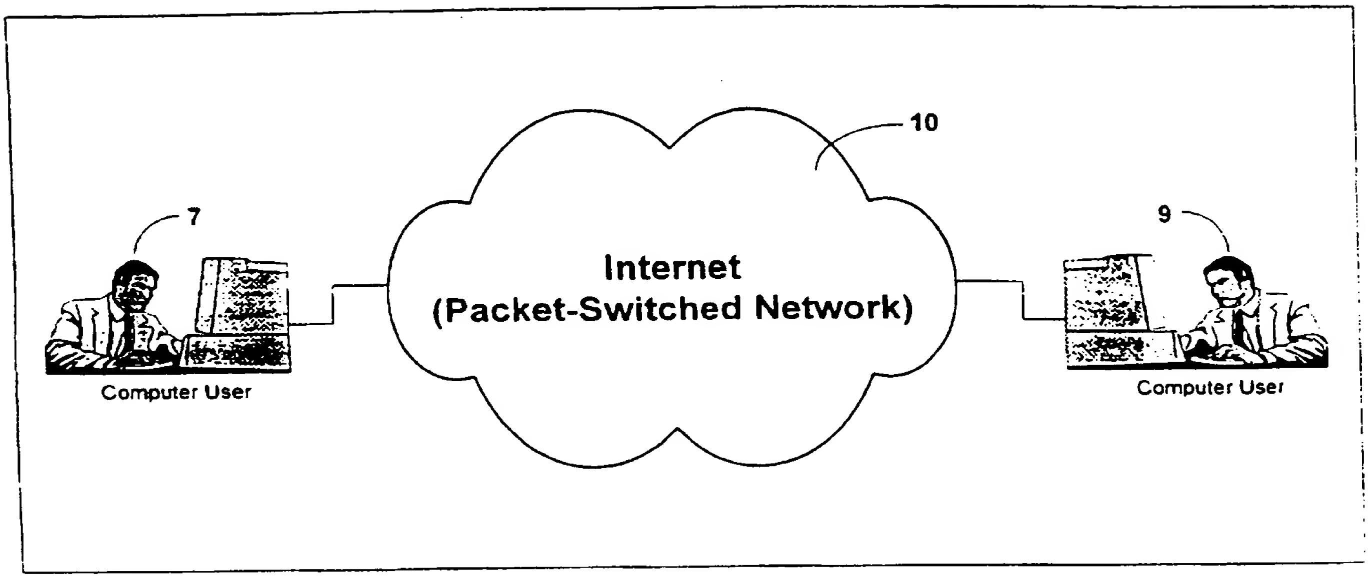


FIG. 2 (PRIOR ART)

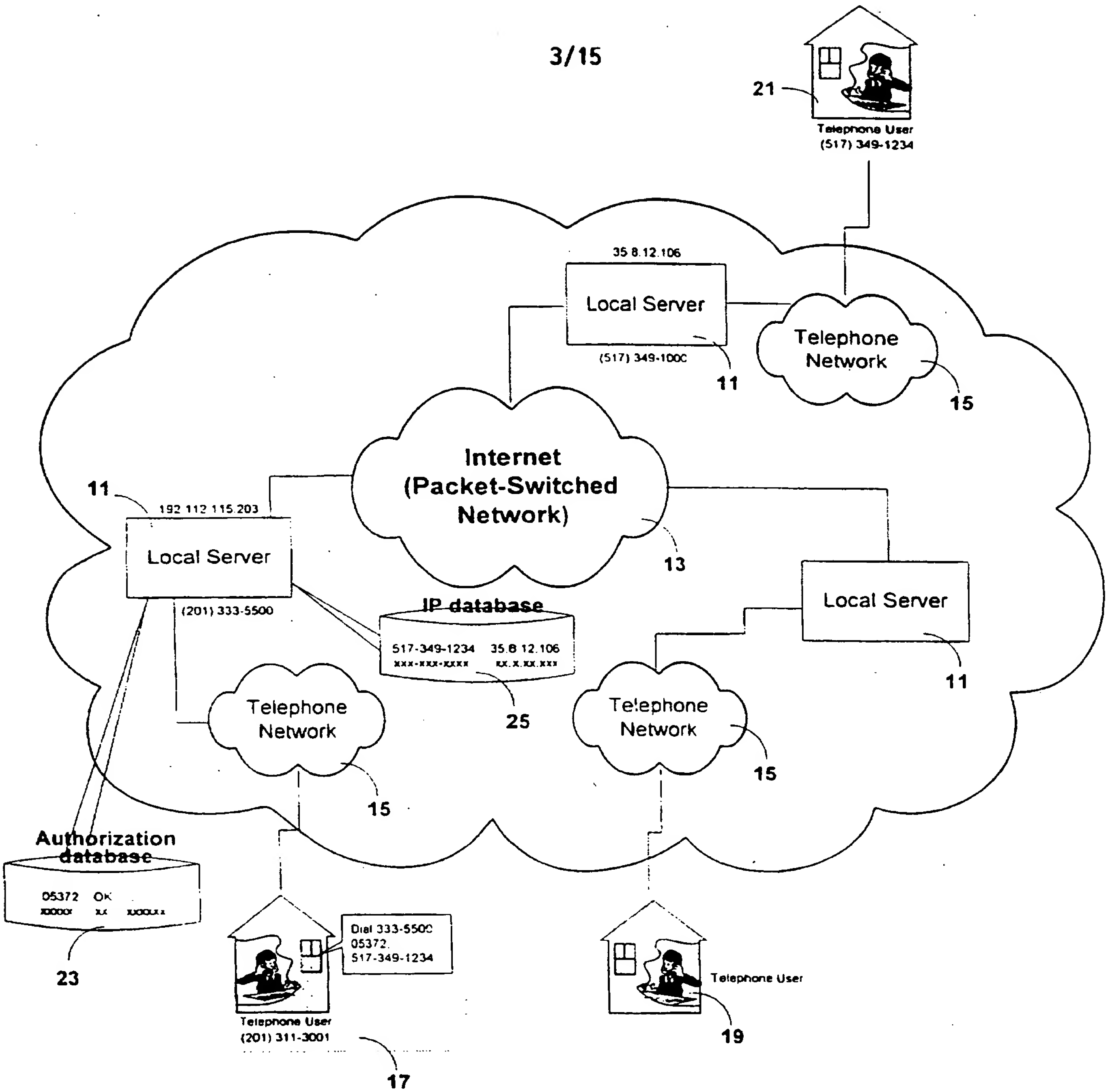

21
Telephone User
(517) 349-1234
35 8.12.106
Local Server
(517) 349-1000
11
Telephone
Network
15
Internet
(Packet-Switched
Network)
13
11
192 112 115.203
Local Server
(201) 333-5500
IP database
517-349-1234 35.8 12.106
25
Local Server
11
Telephone
Network
15
Telephone
Network
15
Authorization
database
05372 OK
23
Dial 333-5500
05372
517-349-1234
Telephone User
(201) 311-3001
17
Telephone User
19

FIG. 3

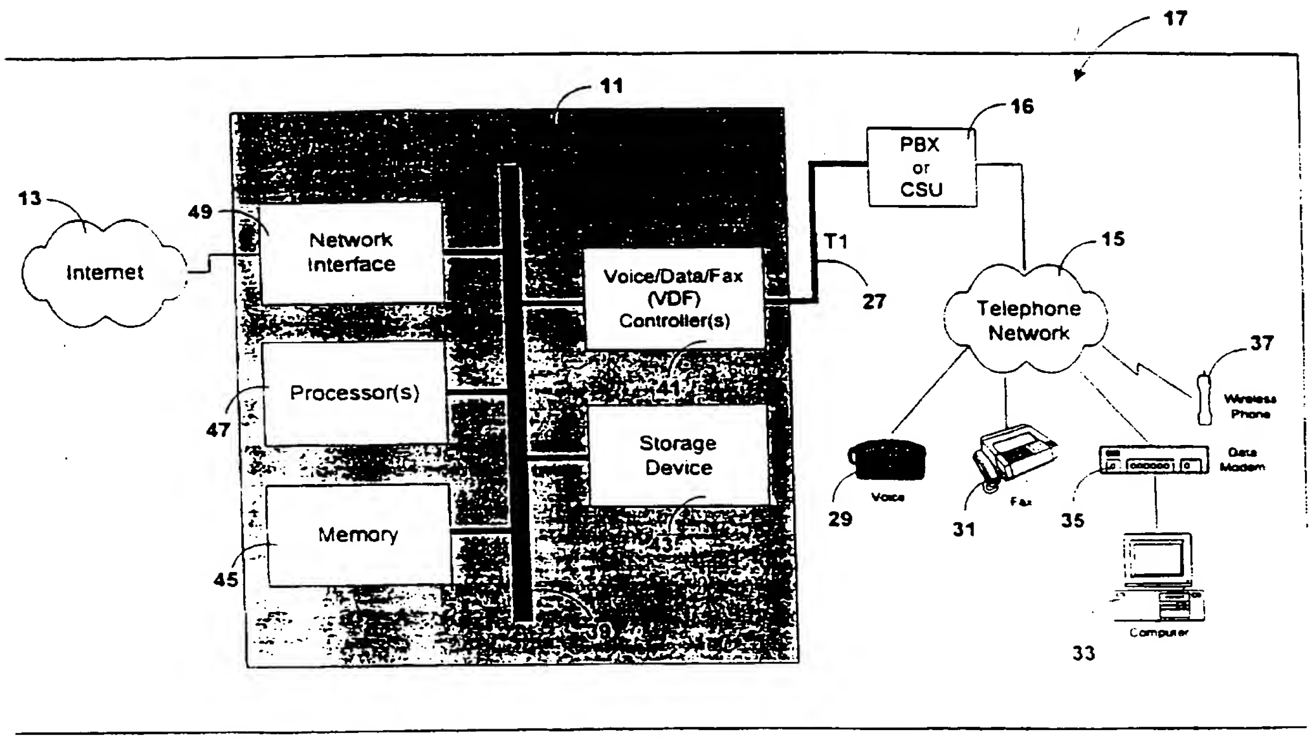

17
11
16
PBX
or
CSU
13
49
Network
Interface
Internet
T1
15
Voice/Data/Fax
(VDF)
Controller(s)
27
Telephone
Network
37
Processor(s)
41
47
Storage
Device
29
31
35
Memory
45
43
39
33

FIG. 4

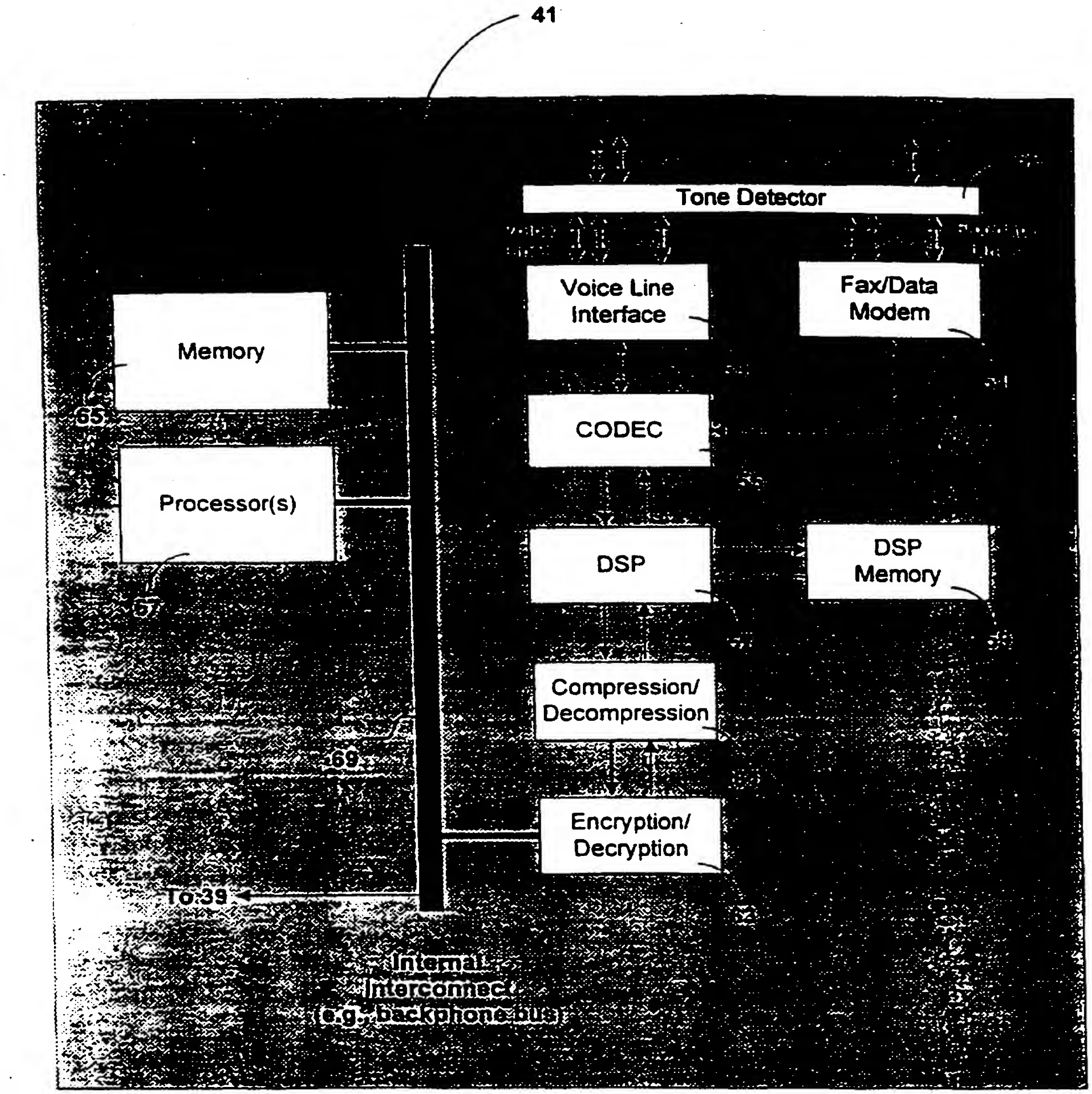

41
Tone Detector
Voice Line Interface
Fax/Data Modem
Memory
65
CODEC
Processor(s)
DSP
DSP Memory
67
Compression/ Decompression
69
Encryption/ Decryption
To 39
Internal Interconnect (e.g., backphone bus)

FIG. 5

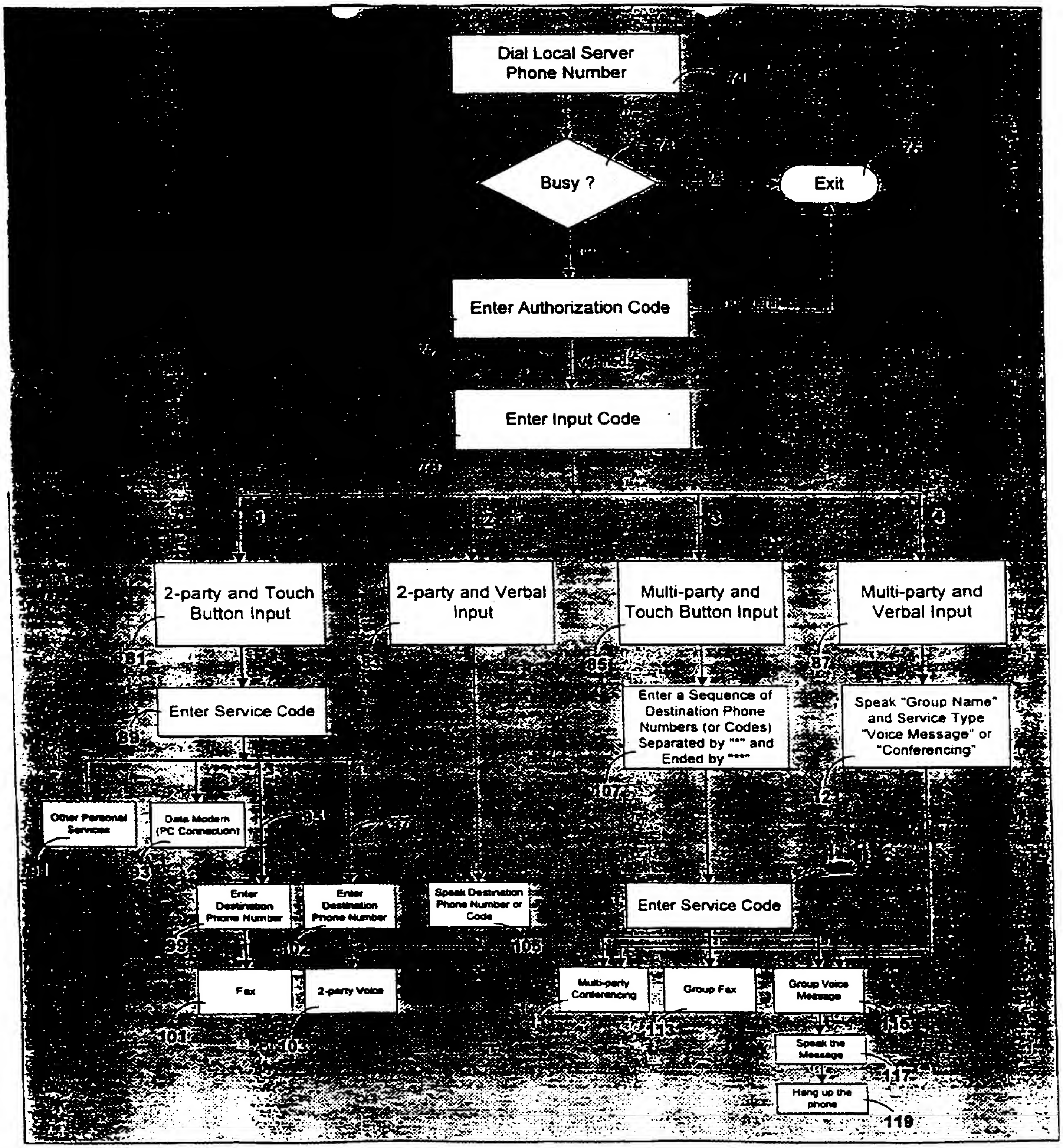

Dial Local Server Phone Number
71
Busy ?
73
Exit
75
Enter Authorization Code
Enter Input Code
1
2
3
4
2-party and Touch Button Input
81
2-party and Verbal Input
Multi-party and Touch Button Input
85
Multi-party and Verbal Input
87
Enter Service Code
89
Enter a Sequence of Destination Phone Numbers (or Codes) Separated by "*" and Ended by "**"
107
Speak "Group Name" and Service Type "Voice Message" or "Conferencing"
Other Personal Services
91
Data Modem (PC Connection)
93
Enter Destination Phone Number
Enter Destination Phone Number
Speak Destination Phone Number or Code
105
Enter Service Code
99
Fax
101
2-party Voice
Multi-party Conferencing
Group Fax
113
Group Voice Message
115
Speak the Message
117
Hang up the phone
119

FIG. 6

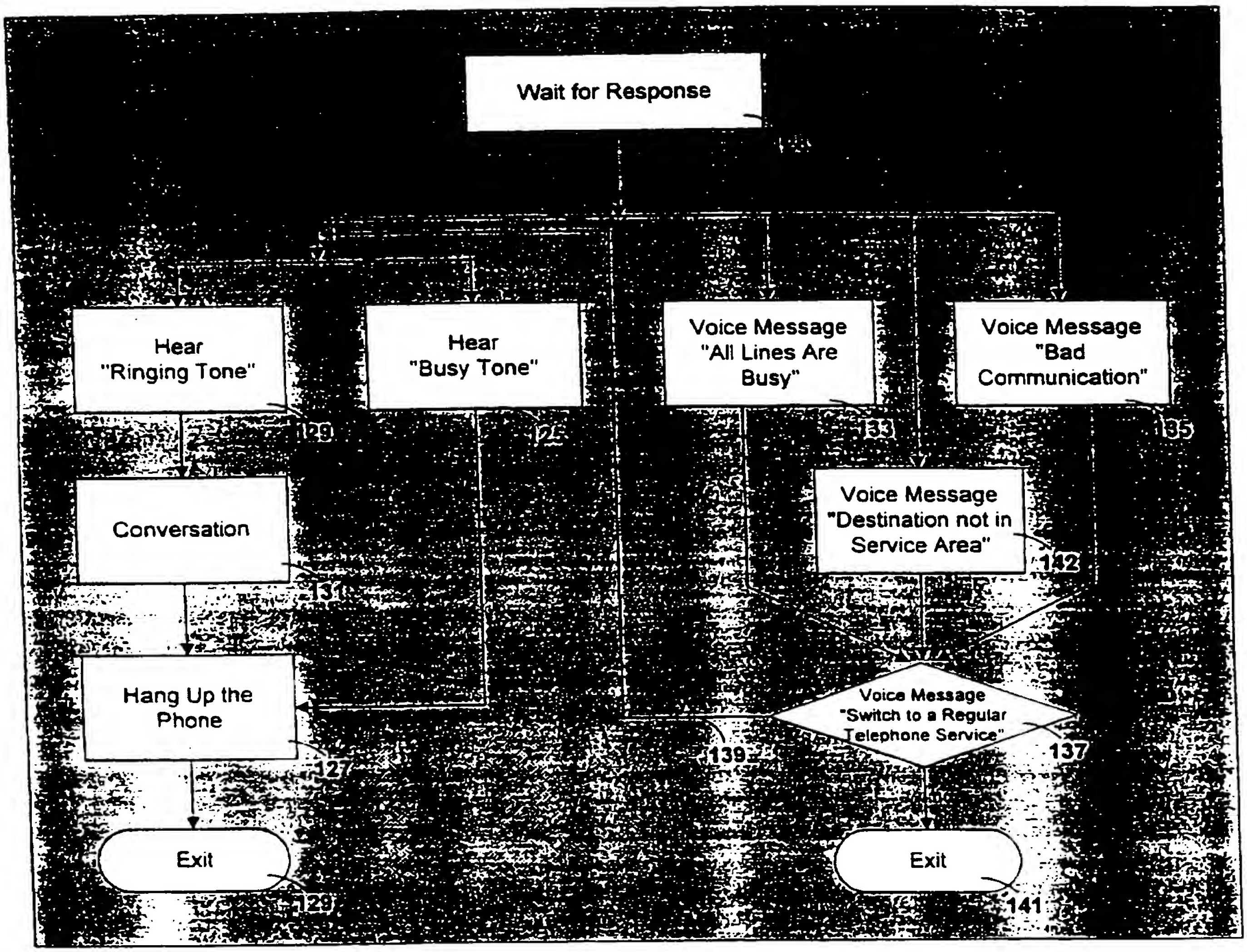

Wait for Response
Hear "Ringing Tone"
Hear "Busy Tone"
Voice Message "All Lines Are Busy"
Voice Message "Bad Communication"
129
125
133
135
Conversation
131
Voice Message "Destination not in Service Area"
132
Hang Up the Phone
127
Voice Message "Switch to a Regular Telephone Service"
139
137
Exit
128
Exit
141

FIG. 7

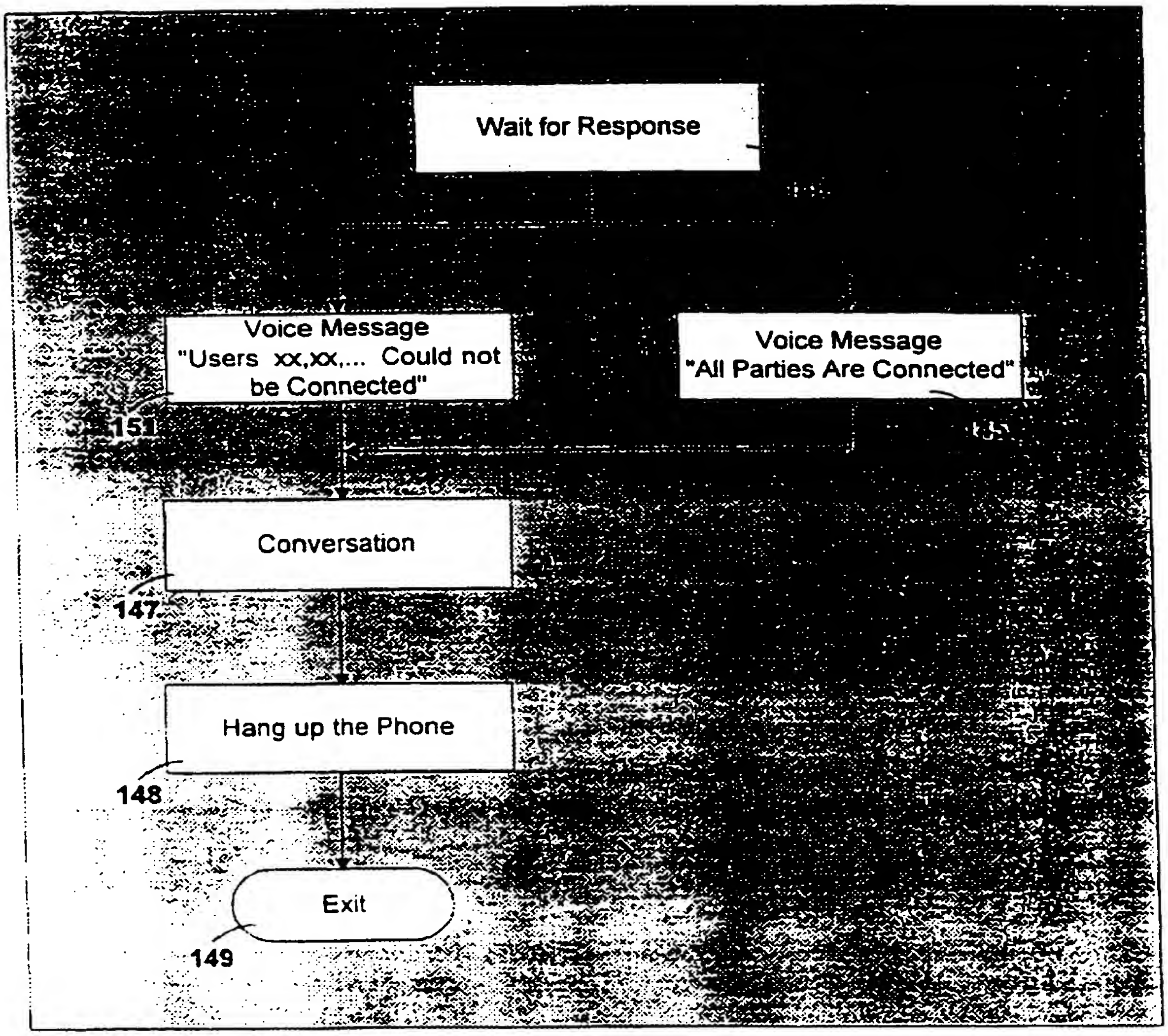

Wait for Response
Voice Message
"Users xx,xx,... Could not be Connected"
151
Voice Message
"All Parties Are Connected"
Conversation
147
Hang up the Phone
148
Exit
149

FIG. 8

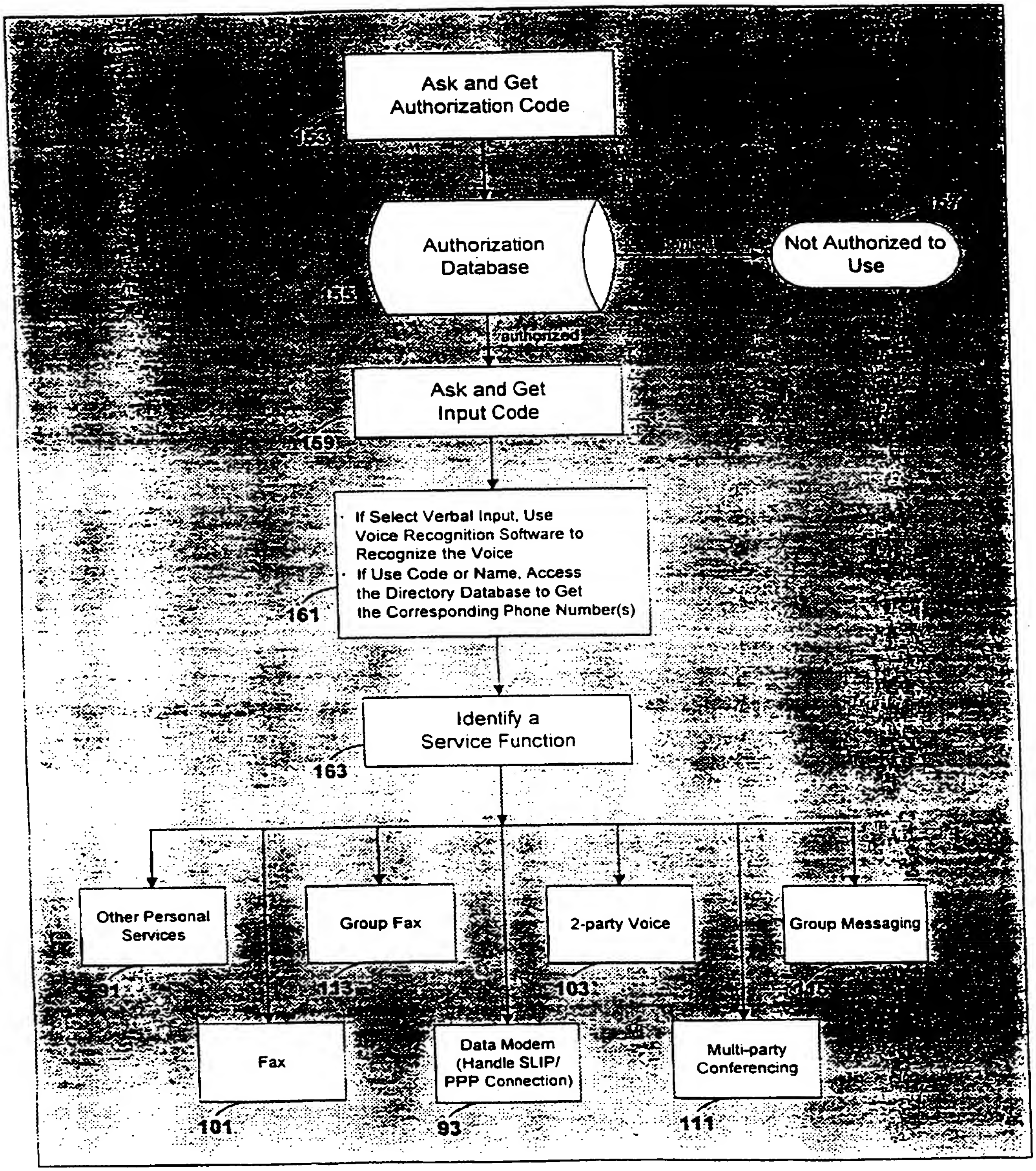

Ask and Get Authorization Code
Authorization Database
155
Not Authorized to Use
authorized
Ask and Get Input Code
159
If Select Verbal Input, Use Voice Recognition Software to Recognize the Voice
If Use Code or Name, Access the Directory Database to Get the Corresponding Phone Number(s)
161
Identify a Service Function
163
Other Personal Services
Group Fax
2-party Voice
Group Messaging
113
103
Fax
Data Modem (Handle SLIP/ PPP Connection)
Multi-party Conferencing
101
93
111

FIG. 9

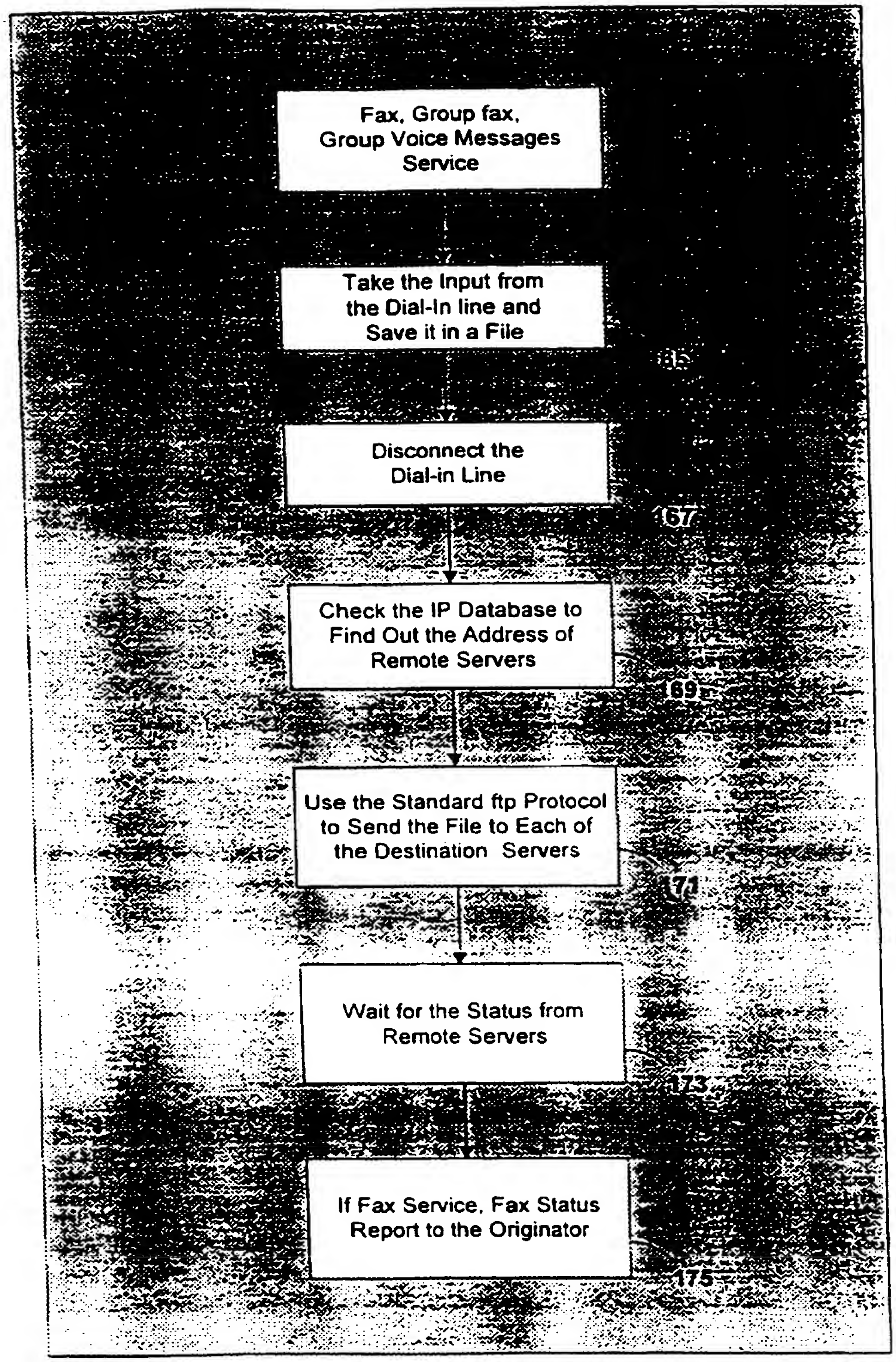

Fax, Group fax, Group Voice Messages Service
Take the Input from the Dial-In line and Save it in a File
Disconnect the Dial-in Line
167
Check the IP Database to Find Out the Address of Remote Servers
169
Use the Standard ftp Protocol to Send the File to Each of the Destination Servers
171
Wait for the Status from Remote Servers
173
If Fax Service, Fax Status Report to the Originator
175

FIG. 10

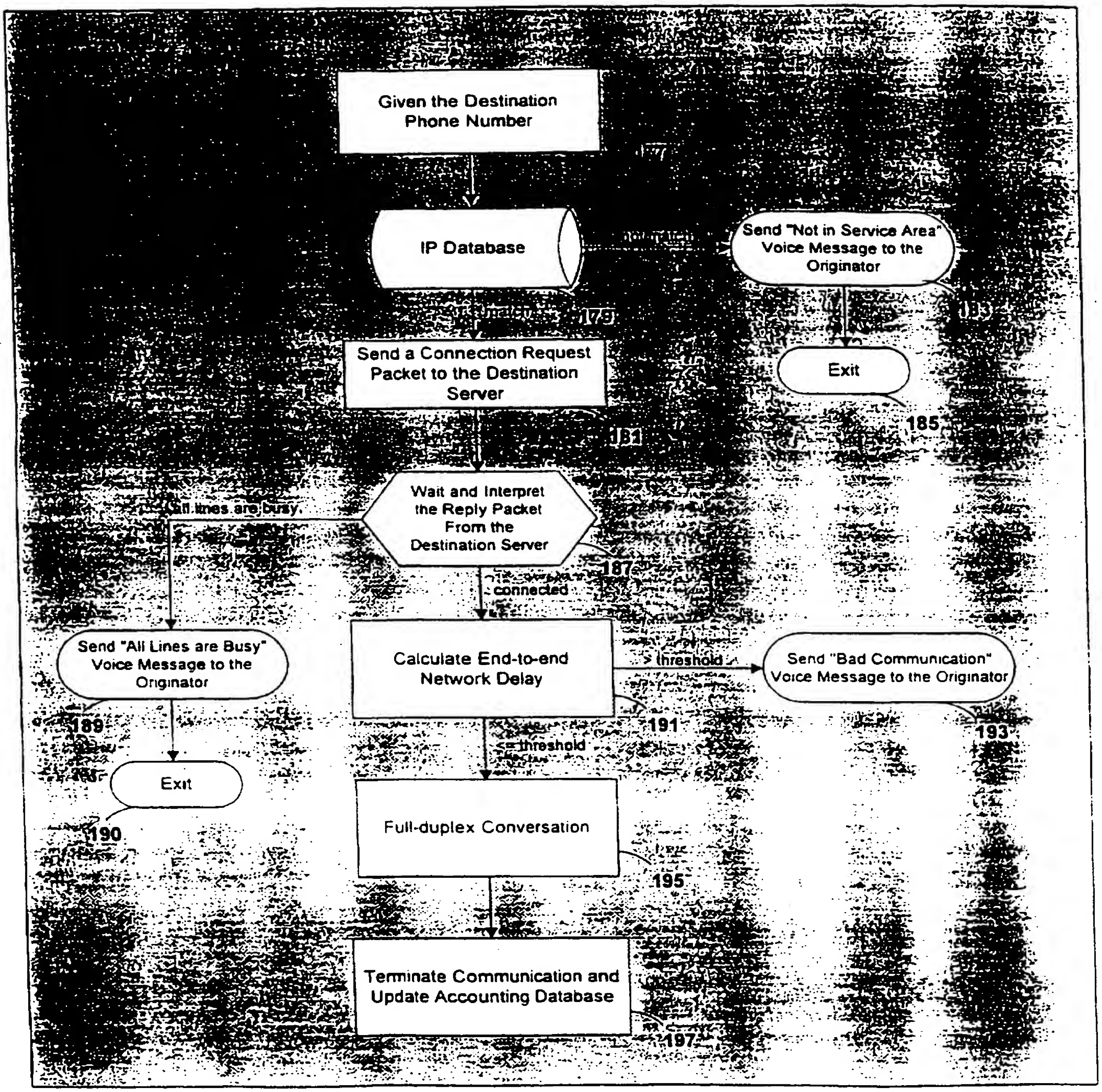

Given the Destination Phone Number
IP Database
Send "Not in Service Area" Voice Message to the Originator
179
Send a Connection Request Packet to the Destination Server
Exit
185
181
Wait and Interpret the Reply Packet From the Destination Server
All lines are busy
187
connected
Send "All Lines are Busy" Voice Message to the Originator
Calculate End-to-end Network Delay
> threshold
Send "Bad Communication" Voice Message to the Originator
189
191
193
<= threshold
Exit
190
Full-duplex Conversation
195
Terminate Communication and Update Accounting Database
197

FIG. 11

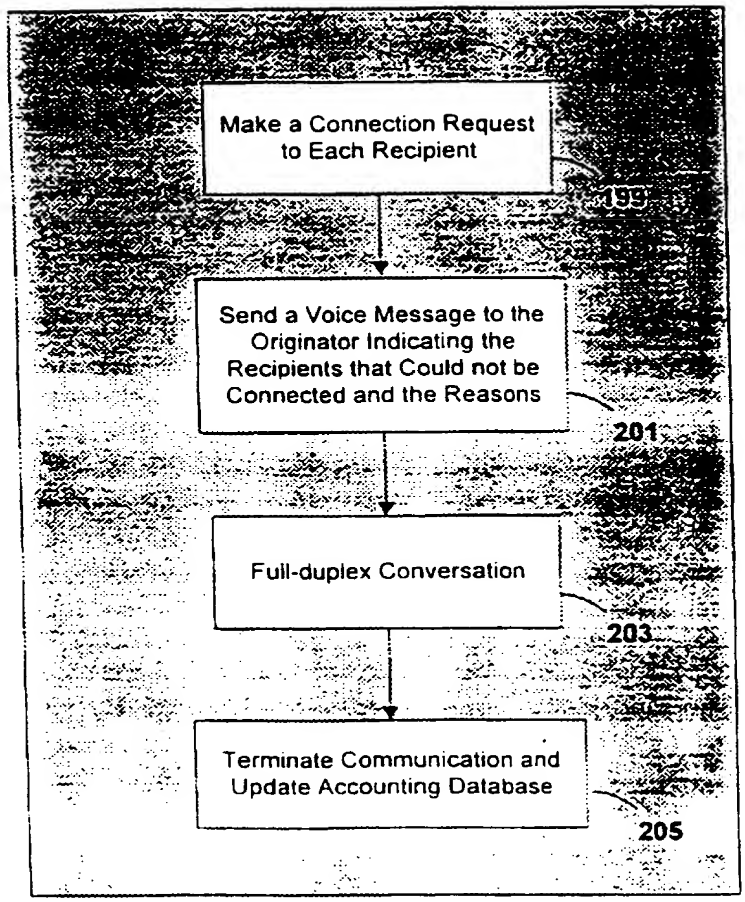
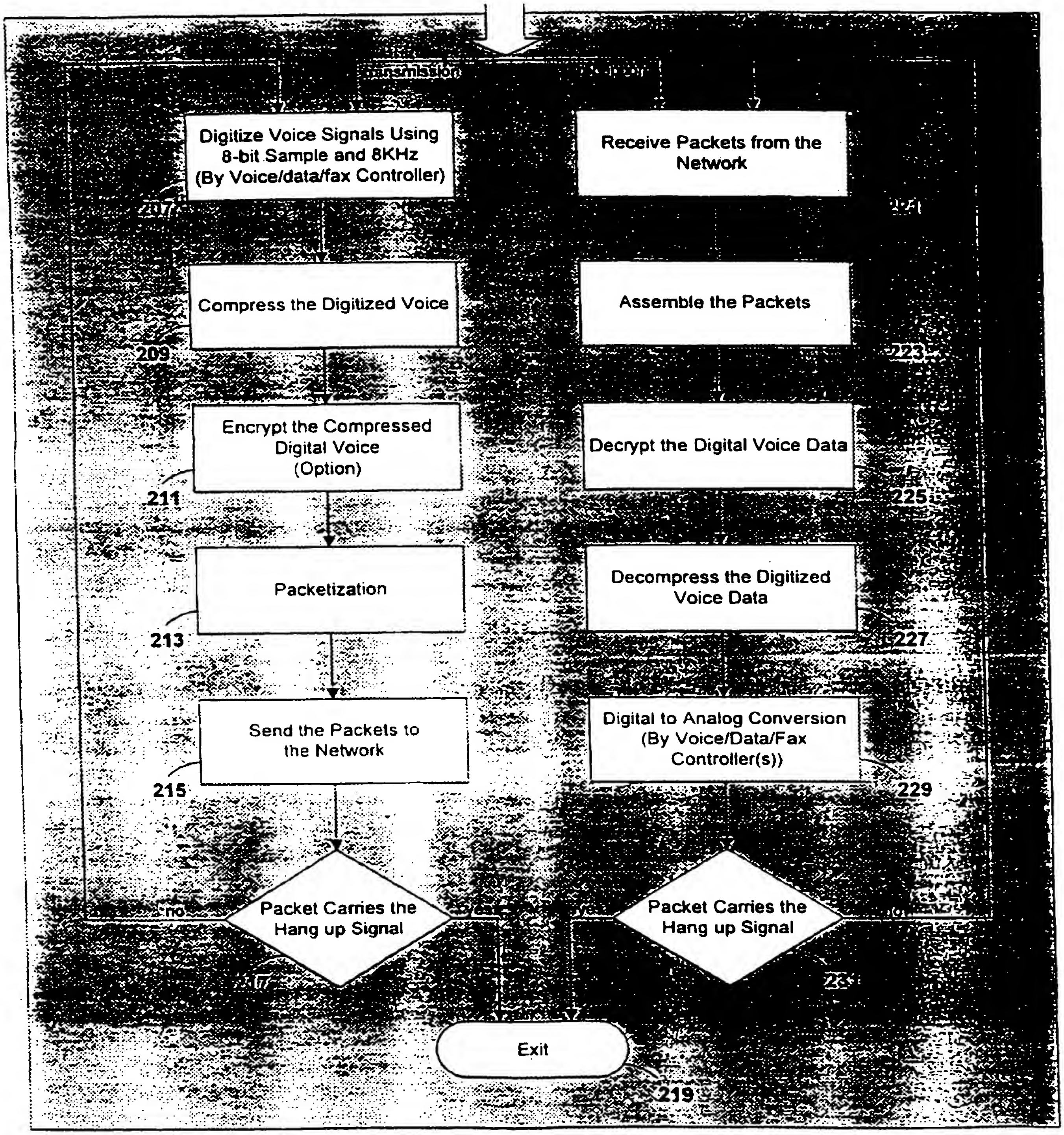

Make a Connection Request to Each Recipient
199
Send a Voice Message to the Originator Indicating the Recipients that Could not be Connected and the Reasons
201
Full-duplex Conversation
203
Terminate Communication and Update Accounting Database
205

FIG. 12



Transmission
Reception
Digitize Voice Signals Using 8-bit Sample and 8KHz (By Voice/data/fax Controller)
207
Compress the Digitized Voice
209
Encrypt the Compressed Digital Voice (Option)
211
Packetization
213
Send the Packets to the Network
215
Packet Carries the Hang up Signal
no
yes
Receive Packets from the Network
221
Assemble the Packets
223
Decrypt the Digital Voice Data
225
Decompress the Digitized Voice Data
227
Digital to Analog Conversion (By Voice/Data/Fax Controller(s))
229
Packet Carries the Hang up Signal
yes
no
Exit
219

FIG. 13

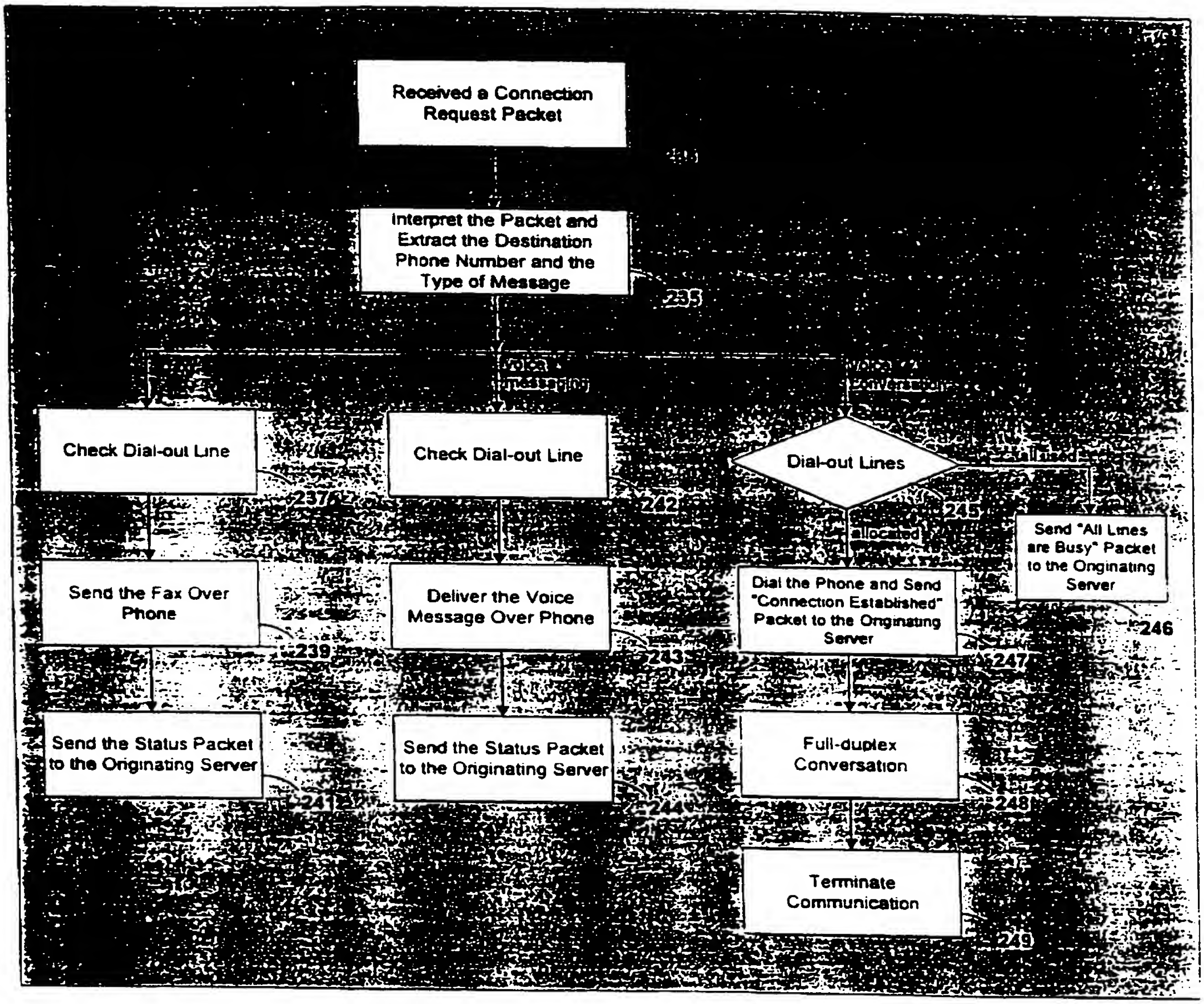

Received a Connection Request Packet
Interpret the Packet and Extract the Destination Phone Number and the Type of Message
235
Voice message(s)
Voice Conversation
Check Dial-out Line
237
Check Dial-out Line
242
Dial-out Lines
all used
245
allocated
Send "All Lines are Busy" Packet to the Originating Server
246
Send the Fax Over Phone
239
Deliver the Voice Message Over Phone
243
Dial the Phone and Send "Connection Established" Packet to the Originating Server
247
Send the Status Packet to the Originating Server
241
Send the Status Packet to the Originating Server
Full-duplex Conversation
248
Terminate Communication
249

FIG. 14

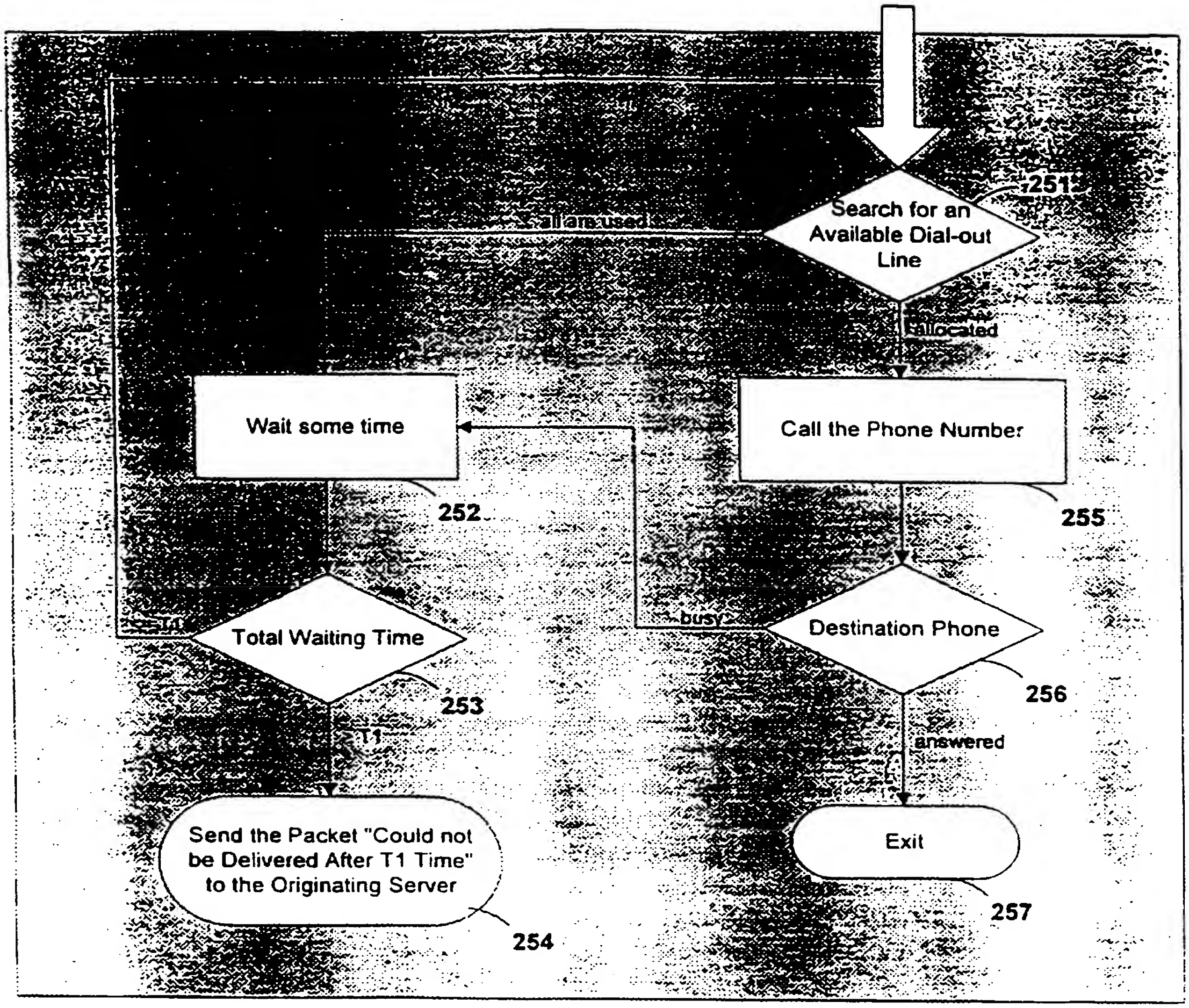

Search for an Available Dial-out Line
251
all are used
allocated
Wait some time
252
Call the Phone Number
255
T1
Total Waiting Time
253
busy
Destination Phone
256
T1
answered
Send the Packet "Could not be Delivered After T1 Time" to the Originating Server
254
Exit
257

FIG. 15

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US97/01589

A. CLASSIFICATION OF SUBJECT MATTER

IPC(6) :H04L 12/56
US CL :370/389
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 370/352, 384, 389, 474, 493; 379/93, 94; 395/200.02

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

APS(packet switched network, digital data network, telephone, server, long distance, originating server, destination server, Internet, interconnecting, Internet phone)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	YANG, C. "INETPhone: Telephone Services and Servers on Internet, Network Working Group", RFP 1789; April 1995, pgs. 1-6.	1-15
Y,E	US 5,608,786 A (GORDON) 04 March 1997, col. 8, line 62 to col. 9, line 17.	1-18
Y	US 4,969,184 A (GORDON et al) 06 November 1990, col. 1, lines 38-56.	1, 4, 9, 15
A,P	US 5,526,353 A (HENLEY et al) 11 June 1996, col. 5, lines 1-30.	1, 4, 9, 15
A	US 5,353,283 A (TSUCHIYA) 04 October 1994.	1, 4, 9, 15
A	US 4,903,261 A (BARAN et al) 20 February 1990.	1, 4, 9, 15

☐ Further documents are listed in the continuation of Box C. ☐ See patent family annex.

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"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

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Date of the actual completion of the international search	Date of mailing of the international search report
19 MARCH 1997	07 APR 1997

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CORRECTED VERSION*

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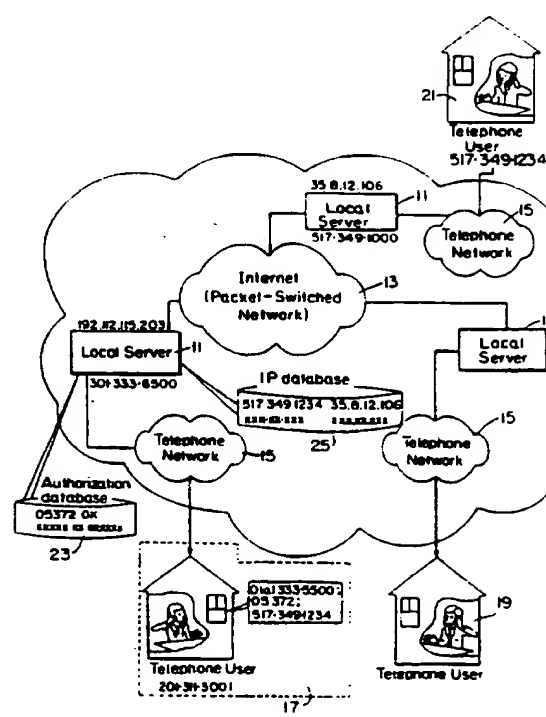
(54) Title: HYBRID NETWORK FOR REAL-TIME PHONE-TO-PHONE VOICE COMMUNICATIONS

(57) Abstract

A method and system are disclosed for permitting regular telephone users (1) to make long distance calls by way of a packet-switched digital data network (13) so as to avoid conventional long distance charges. Personal computers (PCs) are not needed, although they too may use the system. The system includes a plurality of geographically spaced servers (11), each associated with, for example, a particular area code. To make a long distance call, a user (17) simply dials the local number of the originating server (11), and optionally an authorization code (23). The user then inputs the telephone number of the recipient party to the originating server (11). The originating server (11) determines which remote server (11) in the system is local to the number being called and communicates with same via the digital data network (13). Thereafter, the addressed remote server (11) dials out the number of the recipient (19) and real-time voice communication is permitted between the caller (17) and the recipient (19). The system also includes other services such as group messaging, group fax, phone-to-PC communication, PC-to-phone communication, and PC-to-PC communication.



* (Referred to in PCT Gazette No. 57/1997, Section II)

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HYBRID NETWORK FOR REAL-TIME PHONE-TO-PHONE VOICE COMMUNICATIONS

This invention relates to a system and corresponding method for permitting real-time telephone communication between parties via a packet-switched digital data network. More particularly, this invention relates to a hybrid communication network which utilizes an existing circuit-switched telephone network and an existing packet-switched network, the hybrid network including a plurality of geographically spaced servers interconnected via the packet-switched network enabling users to make "long distance" telephone calls by simply accessing their local server, which in turn automatically accesses another server local to the number being called and connects the calling and called parties.

BACKGROUND OF THE INVENTION

Figure 1 illustrates a conventional dedicated telephone network wherein "long distance" calls may be made from for example caller 1 to recipient 3 via network 5. Well known examples of such networks 5 are currently provided by AT&T™ and MCI™ as part of the Public Switched Telephone Network (PSTN). The switching technique of network 5 is based on circuit switching, i.e. each communication is afforded a "dedicated" channel for the duration of the communication. Because caller 1 and recipient 3 are located in different area codes, long distance charges are incurred by the caller upon long distance use of network 5. Unfortunately, these long distance charges quickly multiply and often become quite burdensome.

Long distance subscriber systems (e.g. see U.S. Pat. No. 4,513,175) competing with such established telephone company long distance systems have gained noteworthy acceptance. Typically, such subscriber systems employ the local switched telephone lines of an established telephone company to connect a subscriber to a

computer. The computer conveys the subscriber's telephone call over a dedicated transmission network to another local area where the call is again reintroduced into the switched telephone network and completed to the location dialed. The user is typically required to enter a seven digit local telephone number to gain access to the computer which controls the long distance dedicated network(s) to be employed. The computer answers the call and indicates that access has been gained by placing a tone or the like upon the line to the user. Upon hearing the tone, the user enters an assigned billing code, and thereafter, dials the area code and telephone number of the remote location which is to be contacted through the system.

Unfortunately, the long distance subscriber systems set forth in U.S. Patent No. 4,513,175 suffer from a number of problems. Firstly, the systems are not equipped to permit facsimile communication, multiparty conference calls, etc. as well as conventional telephone conversations. Secondly, these privately owned long distance networks are not packet-switched, and therefore suffer from the problems inherent in dedicated systems. Furthermore, the subscriber systems discussed in the '175 patent require the construction and maintenance of privately owned dedicated transmission media or lines. This is impractical and unduly expensive given today's marketplace.

U.S. Patent No. 5,341,374 discloses a token ring network integrating voice data and video with distributed call processing in a packet-switched network which supports real-time voice conversation. A plurality of token ring networks are interconnected via bridges or the like, each token ring network including a number of node coupling units (processor-controlled switches) arranged in a ring connected by a twisted pair. Each node may be coupled to a PC, telephone, and/or imaging system. Analog-to-digital (A/D) and digital-to-analog (D/A) conversion as well as data processing, display, and storage operations are

performed by the household devices (e.g. telephone, PC, etc.) coupled to the nodes. Unfortunately, the system of the '374 patent is not able to serve the majority of today's society because most households do not own PCs, facsimile machines, and digital telephones which perform A/D and D/A conversion in a MU-LAN PCM format. Households having just simple telephones which output analog voice signals during conversations cannot benefit from or utilize the '374 system. Also, phones not connected to the token-ring local area network (LAN) cannot use the system, i.e. the system is limited to token ring network technology which is undesirable given current market conditions.

U.S. Patent No. 4,969,184 discloses a data transmission system which utilizes a local public switched telephone network (PSTN) in transmitting information between remote data devices by way of a nationwide digital data network. A plurality of geographically spaced local nodes (each connected to a local PSTN) are connected via the digital data network enabling facsimile data, for example, to be transmitted from one area code to another via the digital data network without incurring long distance charges. Unfortunately, the system set forth in the '184 patent has numerous drawbacks, including (i) it is not capable of transmitting real-time continuous voice data; (ii) it requires the use of broadcast facilities and appears to be limited to facsimile or data transmission as opposed to voice transmission; and (iii) it requires the provision and maintenance of a private or paid-for digital data network.

In the last decade or so, the packet-switched digital data network commonly known as the "Internet" has gained popularity throughout the world. Figure 2 illustrates computer 7 communicating with computer 9 by way of the Internet 10. The Internet, the most known world-wide packet-switched network, is a collection of thousands of computer networks, tens of thousands of computers, and more than ten million users who share a

compatible means for interacting with one another to exchange digital data. The system is composed of many network providers interconnected via routers. The most commonly used method for transferring files is known as the file transfer protocol (ftp). Computers 7 and 9 typically access the Internet via various standard network interface cards, such as Ethernet and FDDI, or indirectly by way of data modems. Wire-type links are generally used.

The Internet is a packet-switched digital data network. Packet switching is a way in which different network segments can share a common transmission media. Rather than send a large block of data over a "dedicated" line directly to the destination computer, a packet switching network breaks the data into small chunks, each chunk being sent along a common transmission line in a "packet" that also contains source and destination information. This allows many packets to flow through the same network, all reaching their appropriate destination. Dedicated network components called packet-switching nodes route these packets from source to destination, using the information contained in the packet itself. After all the packets from a particular transmission of data reach their destination, the source and destination information is removed, and the packets are reassembled into the original data. In this way, packets from any number of computers can share the network.

Although it is currently unclear whether the following are prior art to the instant invention, systems which allow computer-to-computer voice communication over the Internet have recently been introduced into the marketplace. Using such systems, voice communication is possible over the Internet provided that the participating computers (PCs) are equipped with their own microphone, speaker, audio device, and necessary communication software. Unfortunately, this recently introduced computer-to-computer voice technology may only be utilized when both parties

have PCs equipped with the specialized hardware, software, and Internet connection. Furthermore, it is required that both parties be pre-notified of intended usage, and both computers be turned on with the necessary software before communication may take place. This is unduly burdensome and impractical as 70% to 80% of the households in the United States do not even have computers, not to mention the even higher percentage of non-computer households throughout the world.

International Discount Telecommunication (IDT) has recently announced a system for providing computer-to-phone voice communication over the Internet. Again, it is unclear at this time whether this system represents prior art to the instant invention. However, this computer-to-phone system also suffers from the problems set forth above regarding the computer-to-computer system and is further limited because it is not bi-directional. In other words, communications or voice conversations can only originate from the PC. Callers who simply own a conventional telephone (i.e. hook and ring device) may not call either PC owners or other phone owners by way of this system. This is a problem.

In view of the above, it is clear that there exists a need in the art for a bi-directional system and corresponding method for permitting real-time voice conversation between telephone users (without the need for PCs or the like) wherein any telephone owner or caller who desires to make a long distance call simply dials a local number which results in real-time voice communication between the caller and recipient via a digital packet-switched network thereby eliminating the incurrence of conventional long distance charges. There also exists a need in the art for such a system which will also support facsimile (fax) transmissions as well as multi-party or conference calls.

SUMMARY OF THE INVENTION

Generally speaking, this invention fulfills the above-described needs in the art by providing a bi-directional telecommunication network enabling real-time voice communication between callers and recipients, the telecommunication network comprising:

a plurality of bi-directional communication servers interconnected by way of a packet-switched digital data network, each of the servers being coupled to users by way of a switched telephone network so that a caller can access an originating server over the telephone network and input a destination telephone number of a recipient; and

wherein each of the servers includes means for receiving one of the destination telephone numbers from a caller and in response establishing real-time voice communication between the caller and the recipient via the destination server over the packet-switched digital data network.

According to certain preferred embodiments, the system also enables facsimile, group facsimile, multi-party voice, and PC-to-PC communication.

This invention further fulfills the above-described needs in the art by providing a method of making a long distance telephone call in real time from a caller to a recipient, the method comprising the steps of:

a) providing a first server local to the caller and a second server local to the recipient, the first and second servers being connected to one another by a digital data network;

b) the caller dialing a local telephone number in order to access the first server by way of a local switched telephone network;

c) the caller selecting a two party voice communication mode from a plurality of possible modes, the other possible modes including a facsimile mode and a PC-to-PC mode;

d) the caller entering the recipient's telephone number which is received by the first server;

e) upon receipt of the recipient's telephone number, the first server instructing the second server via the digital data network to call the recipient;

f) the second server calling the recipient's telephone number by way of a local call in order to connect the caller and recipient via the first and second servers and the digital data network; and

g) the caller and recipient carrying on a real-time voice telephone conversation during which the first and the second servers each perform D/A and A/D conversion of voice signals thereby enabling the parties to carry on the conversation using telephones which output analog voice signals.

In addition to phone-to-phone communication, the system also permits phone-to-PC, PC-to-phone, and PC-to-PC communications, provided that the PCs have an audio device, speaker, microphone, and software to implement same.

This invention will now be described with respect to certain embodiments thereof, accompanied by certain illustrations wherein:

IN THE DRAWINGS

Figure 1 is a prior art illustration of a conventional PSTN system permitting long distance telephone calls between a caller and recipient.

Figure 2 is a prior art illustration of a pair of computers communicating with one another via a packet-switched digital data network such as the Internet.

Figure 3 is a block diagram of a hybrid communication network utilizing existing telephone networks and an existing packet-switched digital data network according to this invention,

the hybrid network including a plurality of geographically diverse bi-directional servers interconnected by the packet-switched network.

Figure 4 is a block diagram illustrating a communication server of the Figure 3 system.

Figure 5 is a block diagram of the voice/data/fax controller of the Figure 4 server.

Figure 6 is a flowchart illustrating how a calling party or caller utilizes the Figures 3-5 system in order to choose between one of multiple different modes of communication.

Figure 7 is a flowchart of the two-party voice mode shown in Figure 6.

Figure 8 is a flowchart illustrating functionality and/or steps associated with the multi-party modes of Figure 6.

Figure 9 is a flowchart illustrating steps carried out by a calling or originating server (i.e. server local to the caller).

Figure 10 is a flowchart of steps carried out by an originating server in facsimile, group facsimile, and group messaging modes.

Figure 11 is a flowchart illustrating steps carried out by an originating server in the two-party voice mode.

Figure 12 is a flowchart illustrating steps carried out by an originating server in the multi-party conferencing mode.

Figure 13 is a flowchart illustrating the functions performed by the servers in the network in both the reception and transmission modes.

Figure 14 is a flowchart illustrating dialing out steps performed by a destination server local to the recipient.

Figure 15 is a flowchart illustrating dialing out functions performed by the called or destination server when real-time communication is not required between the caller and recipient.

DETAILED DESCRIPTION OF CERTAIN EMBODIMENTS OF THIS INVENTION

Referring now more particularly to the accompanying drawings in which like reference numerals indicate like parts throughout the several views.

Figure 3 illustrates a hybrid network for providing real-time telephone voice communication between remotely located callers and recipients, the hybrid network utilizing existing circuit-switched telephone network(s) 15 having dedicated lines and existing packet-switched digital data network 13 (e.g. the Internet). The hybrid network permits callers 17, 19, or 21 having simple telephones (and not a PC or facsimile machine) to make what would otherwise be long distance telephone calls to respective recipients without incurring the conventional long distance charge. The network uses no centralized control and combines the advantages of an existing local telephone network(s) 15 for cost effective local communication with the existence of, for example, the Internet 13 for economic global communication thereby allowing long distance telephone calls to be made without the usual "long distance" expense incurred when the PSTN is used. The hybrid network does not require callers and/or recipients of calls to have any "special" telecommunications equipment such as PCs, faxes, etc. other than a conventional analog-output telephone.

A caller accesses an originating server 11 using a local seven-digit telephone number and enters a recipient's number (destination telephone number including at least ten digits). The originating server looks up the destination number in its IP database 25 and determines the address of the corresponding server 11 local to the destination number (e.g. in the same area code). The originating server 11 then addresses and communicates with the destination server 11 over network 13,

which in turn calls the recipient over PSTN 15. When the recipient's telephone rings, the recipient simply picks up the phone and proceeds to conduct a regular phone conversation with the caller. In the case of voice messaging or multi-party conferencing, the recipient is notified of the type of service (or mode) by way of a voice message sent to the recipient from the destination server. In the case of a fax or group fax modes to be discussed below, the recipient is assumed to have a fax machine.

As shown in Figure 3, the hybrid network includes a plurality of geographically spaced communication servers 11 interconnected by way of packet-switched digital data network 13. According to certain embodiments, each server 11 is located in a different area code or local calling region. For example, Figure 3 illustrates the phone number of the server 11 local user 17 as (201) 333-5500 and the phone number of the server 11 local user 21 as (517) 349-1000. All servers 11 (e.g. PC-based including a Pentium™ processor) function as bi-directional interface devices between digital data network 13 and the switched telephone network 15 in that any one of households 17, 19, and 21 can communicate with one another no matter who originates the communication.

Figure 4 is a block diagram illustrating one of the plurality of servers 11 in detail. Each server 11 is connected to a corresponding local telephone network 15 by way of a private branch exchange (PBX) 16 so that a multiplicity of potential callers/recipients can access the system via each server. Alternatively, a channel service unit (CSU) may be used instead of PBX 16 to permit communication between network 15 and server 11. A standard T1 link 27 may be interposed between PBX 16 and server 11.

As shown, each household (17, 19, or 21) includes at least a standard analog-output telephone 29. Optionally, each household may also include a facsimile machine 31, personal computer (PC) 33, data modem 35, and/or wireless or cellular telephone 37. Each one of these devices may be used to access the hybrid network via an originating server 11 and the proximate local telephone network 15. If the user's phone 29 or PC 33 is equipped with a video display and/or camera, the system is able to support real-time audio/video conversation and imaging between callers and recipients.

Each server 11 includes buss or busses 39 which interconnects voice/data/fax controller(s) 41, storage 43, memory 45, processor(s) 47, and digital data network interface 49. Network interface 49 may be, for example, a conventional Ethernet or FDDI network access card. Multiple network adapter cards may be used when server 11 services many lines, the number of access cards required also being a function of the network bandwidth. Packetized data to be sent over network 13 may be formatted at 49 by way of conventional TCP/UDP/IP based protocols. For real-time voice communication, an efficient low-overhead UDP-based protocol is used. Optionally, the RTP (real-time transport protocol) or the public domain real-time audio transport protocol, vat, slightly modified, may be used.

Digital data storage 43 may include a standard storage disk while a Pentium-based chip(s) may be used in processor(s) 47. Storage 43 includes both authorization database 23 and IP database 25, as well as a directory database. Thus, information relating to which server 11 in the network (and its address) covers, or is local to, particular destination telephone numbers is stored at 43. For example, each server 11 in its storage 43 may include information indicating that if destination telephone number (517) 349-1234 is entered by a caller, then the network 13 address of the server 11 local to that particular number is

35.8.12.106 (see the telephone numbers and addresses shown in Figure 3). Additionally, storage 43 may be used to store accounting information, authorization code data, credit card information, and billing information relevant to particular users or households. Authorization database 23 maintains the authorization codes of active local users and their corresponding credit information. Meanwhile, memory 45 is utilized to store operating or application software used for controlling each server 11 by way of processor(s) 47. Additionally, data retrieved from storage 43 may be temporarily stored in memory 45 while calls and connections are being made.

The directory database within storage 43 maintains the personal directory of each user local to that server 11 (active and past users). For each user, the personal directory includes information such as the name and code of each group and individual which may be called in modes 85 and 87, personal usage information, personal billing information, transaction dates, etc. Because the directory database maintains records of both active and past users, when a past user wants to reactivate their account, the information is easily retrieved. According to certain embodiments of this invention, when a user moves from one area to another, the user's database information at 43 will be automatically transferred over network 13 from one server 11 to another server 11 local the new area, the transferring taking place either at the request of the user or when the user accesses his new originating server 11 for the first time.

By way of each user's personal directory database at the local server 11, the system according to this invention provides the following telephone services: (i) the user may check and delete voice messages left by others in his database; (ii) the user may check the status of group voice messages and faxes previously requested; (iii) directory information - the user may request a telephone number of a particular individual(s) if the

user inputs a name and location; (iv) the user may monitor his personal account, usage, etc.; and (v) the user can create, delete, and modify group names, codes and phone numbers relating to group and individual modes.

The duties or functions of processor(s) 47 include controlling the flow of data packets from controller 41 to network interface 49 and vice versa. Processor(s) 47 also controls the updating, retrieving, etc. of the billing data and the like stored at 43.

Voice/data/fax controller 41, provided in each server 11, is shown in more detail in Figure 5. Controller 41 includes fax/data modem 51, voice line interface 53, coder/decoder (CODEC) 55, digital signal processing unit (DSP) 57, DSP memory 59, compression/decompression device 61, encryption/decryption device 63, memory 65, and optionally processor(s) 67. The various devices shown in Figure 5 which make up each controller 41 are interconnected by way of buss 69 which communicates with buss 39.

Voice line interface 53 and fax/data modem 51 are connected to tone detector 52 which receives and properly distributes voice and/or fax/data signals which are incoming from PBX 16 over link 27. Accordingly, interface 53 receives from tone detector 52 incoming voice signals while modem 51 receives incoming fax/data signals. The detector 52 in controller 41 may be interfaced to the local switched (dedicated) telephone network 15 by way of a loop start (e.g. RJ 11 and/or RJ 14) when only a few voice lines are to be employed, while a standard T1 trunk 27 may be utilized for a larger number of lines (PBX 16 may be needed to distribute calls from the telephone network to an available line depending upon the number of lines being served). Each line can support both dial-in and dial-out functions (voice and/or fax) controlled by the voice processing board (see below).

CODEC 55 (e.g. Motorola MC145480 chip) performs standard analog-to-digital (A/D) and digital-to-analog (D/A) conversion. CODEC 55 functions to convert the analog signals received from interface 53 and/or modem 51 to digital signals (e.g. during a telephone conversation when the caller is outputting analog voice signals to the server via network 15).

On the other hand, because each server 11 is a bi-directional interface, when CODEC 55 receives digital data (e.g. digital voice data) from DSP 57, the CODEC converts it to analog, and thereafter forwards it to the local caller/recipient via either modem 51 or interface 53. Thus, CODEC 55 in each server 11 performs at least the following two functions: (i) converts analog signals incoming from its local caller/recipient to digital signals and forwards same over network 13 to the other party; and (ii) receives digital signals from the other party over network 13, converts the digital signals to analog signals, and forwards same to the local caller/recipient over the telephone network 15.

DSP 57 (e.g. TI TMS320 DSP family) performs sampling to voice grade frequency (e.g. 8 kHz) and may apply forward error correction (FEC) to the digital signals received from CODEC 55 in certain embodiments. DSP 57 performs digital echo cancellation and fax signal demodulation/modulation. DSP 57 also performs compression of the digital data to a lower number of bits (e.g. eight) per sample. In the other direction, DSP 57 functions to decode the error correction and decompress the digital data received from compression/decompression unit 61. DSP memory 59 stores information used in the error correction and compression/decompression processes performed by DSP 57.

Compression/decompression unit 61 performs a different type of compression/decompression than that done by DSP 57, thereby compressing data going out over network 13 and decompressing data coming from network 13. For example, unit 61 may utilize the

known GSM compression/decompression algorithm (about a 5 to 1 ratio). When security is of concern, encryption/decryption device 63 is provided and functions in a known manner (any standard encryption/decryption algorithm such as DES may be used) to encrypt digital data going out over network 13 and decrypt incoming digital data.

According to certain alternative embodiments of this invention, a Dialogic D/240SC-T1 24-port voice processing and T1 board may be utilized (this board including voice input, CODEC, DSP, DSP memory, and T1 connection) in conjunction with a Dialogic FAX/120 12-port fax board (including a fax modem and a fax data CODEC) to make up the above-listed components of controller 41. The Dialogic product supports half-duplex communication. A full duplex product, e.g. Calian VM200 high integration compressed voice/fax module, supports one port and performs the functions of steps 51, 52, 53, 55, 57, 59, and 61.

Processor(s) 67 is optional in that if provided, it controls the operation of the components shown in Figure 5 and the data flow therebetween. On the other hand, processor 67 is not required because processor(s) 47 (see Figure 4) may be utilized to perform these functions.

Beginning with Figure 6, certain embodiments of this invention will now be described by way of a call from a calling party (caller) to a receiving party (recipient) using the system of Figures 3-5. For the purpose of this description, let us assume that caller 17 (telephone number (201) 311-3001) wishes to telephone recipient 21 in a different area code at destination telephone number (517) 349-1234. In step 71, caller 17 begins the process by dialing the local telephone number (333-5500) of the proximate server 11 (originating server) so as to access the server by way of the local telephone network 15. At step 73, it is determined whether or not the local server number is busy. If so, the call is not made and the exit function 75 is performed.

However, if the connection between caller 17 and originating server 11 is made, the caller is prompted to enter an authorization code at 77. The caller may input the authorization code by way of DTMF signals or alternatively in a verbal manner. If the entered authorization code is verified, the caller is prompted to enter an input code at 79 for the purpose of selecting one of a plurality of possible different modes. If the authorization code is not verified, the exit function 75 is performed and the call terminated.

By entering the input code at 79, caller 17 may select one of the four different modes shown in Figure 6, namely, two-party DTMF input mode 81, two-party verbal input mode 83, multi-party DTMF input mode 85, and multi-party verbal input mode 87. The input code entered at 79 may be either verbal or DTMF when caller 17 is using a telephone.

When mode 81 is selected, the caller is prompted to enter a service code at 89 for the purpose of choosing one of the following four modes: i) miscellaneous personal services 91, such as personal directory information stored in the directory data base; ii) data modem mode 93 for PC-to-PC connection over network 13; iii) facsimile transmission mode 95; and iv) two-party real-time voice conversation mode 97. DTMF signals are used at 89 to select one of these modes when caller 17 is using telephone 29 or 37. In fax modes, DTMF inputs may be used at 79 and/or 89, while in PC-to-PC mode 93, the caller may prepend the authorization 77 and input 79 digits as a prefix to the telephone number of the originating server (these digits, once prepared, are saved in a file for automatic dialing).

When caller 17 wishes to utilize his PC in communicating with the recipient's PC, mode 93 is selected. Mode 95 is selected when the caller wishes to send a facsimile transmission to the recipient, while mode 97 is selected via DTMF when the caller wishes to engage in a real-time verbal phone conversation

with the recipient. When fax mode 95 is chosen at 89, caller 17 is prompted at 99 by the originating server 11 to enter the destination phone number of the recipient (e.g. (517) 349-1234), the use of this particular number assuming that the recipient's number is the same for both receiving fax and voice signals. Following step 99, the facsimile connection may be made and the fax sent at 101. Mode 93 also encompasses phone-to-PC and PC-to-phone communication in that caller 17 having a simple analog output telephone 29 may communicate in a real-time voice manner with a recipient having a PC equipped with audio receiving equipment, and vice versa, the PC having an address on packet-switched network 13. When, for example, caller 17 has a telephone and recipient 21 has such a PC, the caller dials the originating server 11 and at the same time inputs to the server (e.g. DTMF) the network 13 address of the recipient's PC. The originating server in turn communicates with the recipient's PC over network 13 thereby enabling real-time voice communication between caller 17 and the user of the PC. In a similar manner, caller 17 may utilize his PC 33 in calling recipient 21 having a simple telephone 29.

When two-party voice conversation mode 97 is chosen at 89, the caller is also prompted to enter the destination phone number (e.g. (517) 349-1234) of the recipient at 102. Thereafter, the destination server 11 local to the recipient is addressed by the originating server 11 via network 13 so that real-time two-party verbal communication may be made between the caller and recipient at 103.

When two-party verbal input mode 83 is chosen at 79, caller 17 is prompted to verbally input the destination phone number of the recipient at 105. Following step 105, the caller and recipient are connected as discussed above. Mode 83 may not be

utilized for facsimile purposes according to certain embodiments of this invention, but could be used in combination with PC mode 93.

When multi-party DTMF mode 85 is chosen at 79, the caller is prompted to enter a sequence of different destination phone numbers via DTMF, each complete telephone number being separated from the others by a "*" DTMF input, and the entire sequence ending with "**" at 107. In other words, the caller inputs a continuous sequence of destination telephone numbers (or codes), each number being separated from the adjacent number by a non-numeric DTMF input (e.g. "*" or "#"). Following the entering of the phone numbers of the parties to be called at 107, caller 17 is prompted to enter a service code at 109 for the purpose of selecting from among the three possible modes shown in Figure 6. Via DTMF, the caller may select from multi-party conferencing mode 111, group facsimile mode 113, and group voice message mode 115.

When multi-party conferencing mode 111 is selected at 109, caller 17 is connected by way of the required destination server(s) 11 to the multiplicity of recipients identified by the sequences entered in step 107 thereby resulting in a multi-party conference call. When mode 113 is selected at 109, the facsimile transmission entered by the caller is automatically sent to the plurality of destinations entered at 107 in a similar manner.

When group voice message mode 115 is selected at 109, a single voice message entered by caller 17 is transmitted to each destination telephone number or recipient identified at 107. In accordance with mode 115, caller 17 speaks the message to be sent at 117 and thereafter hangs up the phone at 119, the spoken message being recorded for later transmission by the originating server 11. After step 119, the originating server 11 determines from database 25 which other servers 11 in the hybrid network need to be contacted in order to communicate with each of the

telephone numbers entered at 107. After communication is made with each recipient, the voice message entered at 117 is sent to all recipients either simultaneously or at different times, depending upon the delay and/or traffic on network 13 (see below).

When multi-party verbal input mode 87 is chosen at 79, the caller is prompted to verbally input the group name and service type (voice message or conferencing) at 121. At 121, caller 17 may, for example, verbally enter the destination numbers of all recipients. Depending upon the input at 121, either mode 111 or 115 is chosen and carried out as discussed above.

It is important that the voice modes 103 and 111 be conducted between the caller and recipient(s) in substantially real-time. However, for voice messaging 115 and fax services 101, 113, which do not have stringent real-time requirements, a conventional file transfer protocol such as "ftp" may be used to transfer the message(s) to the destination server(s) at a time convenient to the servers and network. After the sending of a fax or voice message, the originating server receives at least one transmission status packet from the destination server(s) within a predetermined period of time (defined by the caller) indicating the status of the fax(es) or the voice message(s). In the case of fax services, the originating server faxes a status report to caller 17. For the case of voice messaging, the status is saved in storage 43 of the originating server in the form of a voice message so that caller 17 can check same at a later time via local switched telephone network 15.

Figure 7 is a flowchart illustrating possible responses to caller 17 using two-party voice mode 103 selected by way of either mode 81 or 83. As shown, following the initial communication between caller 17 and the destination server 11 via network 13, the caller waits for a response at 123. If a busy tone is heard 125, the caller simply hangs up the phone 127. On

the other hand, when the caller hears a ringing tone 129, a real-time verbal or voice conversation takes place at 131 between caller 17 and recipient 21 upon the recipient pickup up his/her phone (a message may be left on an answering machine if the recipient does not answer). Following conversation 131, each party simply hangs up the phone 127 and the exit function 129 is performed terminating the call. According to certain embodiments, simply the caller hanging up his phone will effect termination of the call.

Complications can arise while caller 17 is waiting for a response at 123. When it is determined by the originating server 11 that all network 13 lines are busy 133, a pre-recorded message is played to the caller indicating that the caller should switch to a regular telephone service, such as AT&T or MCI (PSTN). When originating server 11 determines that there is a bad communication via either network 13 or the remote telephone network 15, at 135, a similar pre-recorded message is played to the calling party advising a switch to conventional telephone service 137. Such a "bad communication" message could, for example, result from a caller-to-recipient network 13 delay which exceeds a predetermined threshold (see Fig. 11). Following the playing of such a message to caller 17 at 137 in response to one of findings 133 and 135, the caller may opt to have the originating server 11 automatically switch the caller to regular long distance service via PSTN 15. If the caller chooses this option, then the call is forwarded at 139 to the recipient's telephone number via the PSTN. If the caller in response to the message at 137 chooses not to be connected via conventional long distance service, then exit function 141 is performed and the call terminated.

Depending upon the number of servers 11 in the hybrid network located throughout the country or throughout the world, it may be the case that the telephone number of the recipient being dialed is not local to a particular server 11 (i.e. the destination number is not found in server database 25). If such is the case, it is determined by the originating server at 142 at which time a pre-recorded message is played to the caller at 137 asking whether or not the caller wishes to be switched to the PSTN as set forth above.

Figure 8 is a flowchart of multi-party conferencing mode 111 as selected by way of either mode 85 or 87. When mode 111 is selected, caller 17 waits for a response at 143. When it is determined by the originating server 11 (i.e. the server local to the calling party) at 145 that all parties identified in either step 107 or 121 are connected, the conference call is begun 147. After the conference call is over, the caller hangs up the phone 148 and the connection is terminated 149. However, when the originating server determines that one or a number of parties identified at 107 or 121 cannot be reached for one reason or another (e.g. line busy or excessive network delay), a voice message is played at 151 to the caller identifying which parties could not be connected. If all parties cannot be reached, caller 17 may simply terminate the call. Otherwise, the conference call may be started at 147 with only the parties which could be reached in attendance. Optionally, according to certain alternative embodiments of this invention, the parties which could not be connected at 151 may be accessed by the originating server 11 via a conventional long distance network (e.g. PSTN) and plugged into the conference call 147 with the parties accessed over the hybrid network.

Figure 9 is a flowchart illustrating the functionality of an originating server 11. As defined herein, an originating server is the server 11 local to and accessed by the calling party (caller). Upon connection between caller 17 and originating server 11, the server at 153 prompts the caller to input an authorization code. Upon receipt of the authorization code (e.g. DTMF), originating server 11 accesses at 155 its authorization database 23, 43, in order to determine if the authorization code is valid (whether it may be verified). When the server 11 determines at 155 that the authorization code input by the caller is improper or invalid, access to the hybrid network is denied at 157. However, if the server 11 determines that the authorization code input by the caller is valid, access to the hybrid network is authorized and originating server 11 prompts the caller at 159 to enter an input code in order to choose between the plurality of possible modes 81, 83, 85, and 87. Following step 159, the caller enters, for example, a DTMF input code (see reference numeral 79 in Figure 6) in order to select a mode of operation. As shown at 161, voice recognition and processing software is utilized when one of modes 83 and 87 is selected. Server 11 looks up in storage 43 (IP database 25) the remote server 11 address on network 13 covering or corresponding to the telephone number of the recipient (i.e. destination number). Select step 163 in Figure 9 encompasses the multiple steps shown in Figure 6 relating to mode selection. For example, steps 89, 107, 109, 121, etc. are included in service type identification step 163. Following step(s) 163, the different functions 91, 93, 101, 103, 111, 113, and 115 may be utilized as described above with respect to Figure 6.

Figure 10 is a flowchart illustrating the steps taken in fax mode 101, group fax mode 113, and group voice message mode 115 in the originating server 11. After one of modes 101, 113, and 115 is selected as shown in Figure 6, the originating server 11

receives the corresponding input from caller 17 by way of line 27 and saves it in either storage 43 or memory 45 at step 165. Thereafter, the dial-in line between caller 17 and server 11 is disconnected at 167. The originating server 11 takes the recipient's telephone number (e.g. (517) 349-1234) input from caller 17 and looks up in IP database 25 the appropriate server 11 which needs to be addressed. For example, as illustrated in Figure 3, the destination server 11 address corresponding to (517) 349-1234 is 35.8.12.106. This takes place at 169.

Following the determination by the originating server as to which server 11 needs to be addressed, the originating server sends file packets to each of the destination server(s) 11 at 171. Thereafter, the destination server(s) dials the recipient's number(s) input at 99, 107, or 121, and connects to the recipient. The originating server waits for a status update from the destination server(s) at 173. For example, when a single or group facsimile transmission is sent, the status is reported to the originating server at 175. Thereafter, caller 17 is free to dial the originating server 11 and determine the status of the fax (i.e. whether or not it was sent).

Figure 11 is a flowchart illustrating the steps taken by originating server 11 when two-party voice mode 103 is chosen by caller 17. Firstly, the server 11 receives and interprets the destination phone number (e.g. (517) 349-1234) entered by the caller at 177 and looks it up in its IP database at 179 to make sure that the hybrid system includes a server 11 local to that destination phone number. If IP database 25 lists a server address covering the received destination phone number (i.e. a match is found), then the originating server sends a connection request packet to the destination server 11 at 181. If the originating server at 179 determines that the hybrid system does not include a server 11 local to or covering the received destination phone number (i.e. no match is found), a voice

message is sent to caller 17 at 183 indicating that the destination phone number is not in the service area of the hybrid network. Thereafter, the call may be terminated 185.

After sending the connection request packet 181, the originating server 11 at 187 receives a reply packet from the destination server 11 indicating that either a connection has been made (or that all lines are busy). When all lines are found to be busy, the originating server sends an appropriate message to caller 17 at 189 and thereafter the call may be terminated 190.

When at 187 the originating server receives a reply packet from the destination server indicating that a connection has been made, the originating server at 191 compares the end-to-end network delay based upon the initial connection with a predetermined delay threshold in order to control the quality of real-time voice conversation. For example, if the predetermined threshold is 1.0 seconds, then it is determined at 191 by the originating server whether the end-to-end delay is greater than, or less than or equal to 1.0 seconds. If the delay is greater than 1.0 seconds (e.g. due to network congestion or the failure of the destination server), then a "bad communication" voice message is sent to caller 17 via telephone network 15 at 193. According to certain embodiments, the originating server 11 gives the caller the option (in the form of a voice message) to automatically dial the destination phone number through the regular PSTN following the "bad communication" message.

When it is determined at 191 that the end to end network delay between the originating server and the destination server is less than or equal to 1.0 seconds, then a full-duplex voice conversation takes place in real-time between caller 17 and recipient 21 at 195. Following the termination of the real-time

telephone call at 197, the length of the telephone call (the time of the call) is recorded in storage 43 so that caller 17 can be billed accordingly.

Figure 12 is a block diagram illustrating the steps taken by the originating server when multi-party conferencing mode 111 is selected by caller 17. At step 199, the server 11 makes a connection request to the requisite destination server(s) covering the destination telephone numbers entered at 107. An efficient multicast protocol such as the IP multicast protocol available on the Internet is used. Thereafter, in step 201, after the connection reply packets have been received, the originating server sends a voice message to caller 17 indicating which, if any, recipients or recipients could not be reached for the reasons discussed relative to Figure 7. At this time, the caller can either begin the multi-party conversation in real-time at 203 with the connected parties or hang up the phone which triggers the termination of all established connections. Following the termination of the multi-party conversation at 205, the originating server updates its billing records for caller 17. The caller is billed accordingly.

Figure 13 is a more detailed flowchart illustrating how a destination server handles a real-time full duplex voice conversation between caller 17 and recipient 21. The steps taken by server 11 transmitting signals over network 13 are illustrated on the left-hand side of Figure 13 while the steps carried out by server 11 in receiving signals over network 13 are illustrated on the right-hand side of Figure 13. In Figure 13, when a server 11 is the transmitting server, CODEC 55 digitizes received voice signals from the caller or recipient at 207. For example, CODEC 55 may utilize 8 KHz sampling and 8-bits per sample so that the controller generates 64K bits per second. Next, after CODEC 55 forwards the digital signal to DSP 57, compression device 61 compresses the digitized voice signal at 209 in order to reduce

network traffic (e.g. GSM compression algorithm). Thereafter, it is optional at 211 to utilize encryption device 63 to encrypt (e.g. DES) the compressed digital voice signal, depending upon whether security is of concern. From encrypter 63, the digitized signal is forwarded by way of buss 39 to network interface 49 where it is placed into a number of packets at 213 for transmission over digital data network 13. It is noted that compression/decompression and encryption/decryption may be performed either by special hardware chips (see Fig. 5) or by software executed by the processor(s) 47 in server 11. Multiple processors 47 may be needed if there are many lines to handle.

Thus, a server 11 acting in its transmitting mode sends the digitized packets at 215 through network 13 to the other server. At step 217, it is determined whether a hang-up signal has been sent (controller 41 is able to detect silent signals and hang-up signals). In the case of silent signals, no packet is sent so as to reduce network traffic. When a hang-up signal is detected, server 11 terminates the connection at 219 to the remote server 11, and thereafter updates the statistic information in storage 43 as to the connection time, called phone number(s), total number of packets transmitted, and the total number of packets dropped by the network.

The originating server 11 may continuously during a communication between a caller and recipient monitor the number of packets dropped or delayed over network 13 and compare the percentage to a predetermined tolerable threshold (e.g. 5%). If it is found that the percentage is greater than the 5% threshold, then a message is sent to the caller indicating that he will not be charged for the call.

Still with reference to Figure 13, we turn to the steps taken by a server 11 in the receiving mode. Firstly, the server receives packet data from network 13 at 221. Thereafter, server 11 assembles the packets at 223 and utilizes decrypting device 63

in order to decrypt the digital voice data at 225. Decompression device 61 then decompresses the digitized voice data at 227 and CODEC 55 converts the digital signal to analog at 229. When it is determined at 231 that a received packet from network 13 includes a hang-up signal, exit function 219 is performed.

Figure 14 is a flowchart illustrating the steps or functions performed by a called or destination server 11. Firstly, at 233, the server receives a connection request packet from an originating server 11 via network 13. The packet is interpreted in order to determine the type of request. When the request relates to a long distance call or the like (Figure 6), the destination phone number is extracted at 235. If the fax mode is selected, the server will try to allocate an available dial-out line 237, send the fax 239, and transmit a status packet back to the originating server at 241.

Meanwhile, when a voice messaging mode is selected, the dial out line is checked at 242. The received voice message is delivered over a dedicated line at 243 following the connection with an available line, and a status packet is sent back to the originating server at 244.

If a voice conversation mode is selected, it is determined at 245 whether dial-out lines are available. If all lines are busy, a message indicating same is sent back to the originating server at 246. If a phone line(s) is available and a connection is made with the destination phone number (e.g. (517) 349-1234), then the destination server at 247 sends a "connection established" packet back to the originating server. Thereafter, a real-time voice conversation takes place 248 and is terminated when desired 249.

For voice messaging and facsimile transmission modes, the real-time constraint is not stringent. Thus, if no dial-out line is available at 237 or 242, the destination server 11 will keep trying within a predetermined time period as shown in Figure 15,

which is a flowchart illustrating the steps performed in the dialing out to the recipient by the destination server 11. Firstly, the server searches for an available dial-out line at 251. When all are found to be in use, the destination server waits a predetermined period of time 252 before again searching for an available dial-out line. After the total waiting time breaks a predetermined threshold 253, the server sends a packet back to the originating server indicating that the connection could not be delivered after a predetermined period of time 254. When an available line is located at 251, the destination phone number (e.g. (517) 349-1234) is called 255. A determination is made at 256 whether the phone of recipient 21 is busy. If busy, the server proceeds to 252 while if answered, the connection is made between the caller and recipient and the routine is exited 257.

Once given the above disclosure, therefore, various other modifications, features, or improvements will become apparent to the skilled artisan. Such other features, modifications and improvements are thus considered a part of this invention, the scope of which is to be determined by the following claims.

WE CLAIM:

1. A method of making a real-time long distance telephone-to-telephone call from a caller to a recipient, the method comprising the steps of:

providing an originating communication server local to the caller;

providing a destination communication server local to the recipient;

interconnecting the originating server and the destination server via a packet-switched digital data network;

the caller telephoning the originating server using a local telephone number via a local switched telephone network, and thereafter communicating to the originating server a destination telephone number of the recipient;

the originating server forwarding to the destination server packetized digital data indicative of the telephone number of the recipient;

the destination server telephoning the recipient using a local telephone number via a switched telephone network and thereafter causing the caller and recipient to be connected for real-time voice conversation;

the originating server converting analog voice signals received from the caller to digital voice signals and thereafter forwarding same to the destination server in packet form via the digital data network during the real-time telephone conversation; and

the destination server receiving the digital voice signals from the originating server and converting same to analog voice signals and forwarding the analog voice signals to the recipient during the telephone conversation.

2. The method of claim 1, further comprising the steps of:

the destination server receiving analog voice signals from the recipient, converting them to digital signals and transmitting same to the originating server in packetized form during the conversation; and

the originating server receiving the packetized digital voice signals from the destination server, converting same to analog, and forwarding the analog voice signals to the caller via the telephone network.

3. The method of claim 1, further comprising the steps of:

determining a network delay between the originating and destination servers;

comparing the delay to a predetermined threshold delay time so that when the delay is less than or equal to the threshold, the conversation is permitted to take place.

4. A hybrid bi-directional telephone communication network for permitting real-time phone-to-phone long distance voice conversation between a first party and a second party, the hybrid network utilizing a circuit-switched telephone network and a packet-switched digital data network, the hybrid network comprising:

a first communication server local to the first party and coupled thereto via a switched telephone network, and a second communication server local to the second party and coupled thereto via a switched telephone network;

a packet-switched digital data network interconnecting and allowing packetized digital data communication between said first and second servers;

said first server including transmission mode means for: (i) receiving a local telephone call from the first party by way of the switched telephone network; (ii) receiving from the first party the telephone number of the second party; (iii) communicating with said second server over said digital data network and instructing said second server to call the telephone number of the second party; and (iv) converting analog voice signals received from the first party to digital signals and forwarding same to said second server in packetized form thereby enabling real-time telephone-to-telephone voice conversation between the first and second parties via the digital data network;

said first server further including receiving mode means for: (v) calling the first party upon receiving instructions to do so from the second server; and (vi) receiving digital voice signals from the second server via the digital data network and converting same to analog voice signals and forwarding the analog voice signals to the first party during the voice conversation;

said second server including transmission mode means for: (i) receiving a local telephone call from the second party by way of the switched telephone network; (ii) receiving from the second party the telephone number of the first party; (iii) communicating with said first server over the digital data network by way of data packets instructing the first server to call the telephone number of the first party; (iv) converting analog voice signals received from the second party to digital voice signals and forwarding same to said first server over the digital data network;

said second server further comprising receiving mode means for: (v) locally calling the second party upon receiving instructions to do so from said first server; and (vi) receiving packetized digital voice signals from said first server over said

digital data network and converting same to analog voice signals and forwarding the analog voice signals to the second party during the voice conversation; and

wherein said hybrid network including said first and second servers is bi-directional in that both the first and second parties are capable of initiating long distance telephone calls to the other using their respective telephones which output analog voice signals.

5. The hybrid network of claim 4, wherein each of said first and second servers further comprises facsimile mode means for allowing the first and second parties to send facsimile transmissions to one another over said digital data network.

6. The hybrid network of claim 5, wherein each of said first and second servers further comprises conference call means and group message means for allowing the first and second parties to conduct conference calls and send group messages respectively over the hybrid network via the digital data network; and

group facsimile means.

7. A method of making a long distance telephone call in real-time from a caller to a recipient, the method comprising the steps of:

a) providing a first server local to the caller and a second server local to the recipient, the first and second servers being connected to one another by a digital data network;

b) the caller dialing a local telephone number in order to access the first server by way of a local switched telephone network;

c) the caller selecting a two-party voice communication mode from a plurality of possible modes, the other possible modes including a facsimile mode and a PC-to-PC mode;

d) the caller entering the recipient's telephone number which is received by the first server;

e) upon receipt of the recipient's telephone number, the first server instructing the second server via the digital data network to call the recipient;

f) the second server calling the recipient's telephone number by way of a local call in order to connect the caller and recipient via the first and second servers and the digital data network; and

g) the caller and recipient carrying on a real-time voice telephone conversation during which the first and second servers each perform D/A and A/D conversation of voice signals thereby enabling the parties to carry on the conversation using telephones which output analog voice signals.

8. The method of claim 7, further comprising the step of determining a delay over the digital data network between the first and second servers, and comparing the delay with a predetermined threshold.

9. A bi-directional telecommunication network enabling real-time voice communication between callers and recipients, the telecommunication network comprising:

a plurality of bi-directional communication servers interconnected by way of a packet-switched digital data network, each of said servers being coupled to users by way of a switched telephone network so that a caller can access a local said server over the telephone network and input a destination telephone number of a recipient; and

wherein each of said servers includes means for receiving one of said destination telephone numbers from a caller and in response establishing real-time voice communication between the caller and the recipient via another said server over said packet-switched digital data network.

10. The network of claim 9, wherein each of said servers further comprises digital-to-analog conversion means for receiving analog voice signals from a local caller or recipient, converting same to digital signals, and thereafter transmitting said digital signals in packetized form over the digital data network to the other of said caller and recipient by way of another said server.

11. The network of claim 10, wherein each of said servers further comprises facsimile means for enabling callers to transmit facsimile data to recipients whereby facsimile transmissions originate from the caller, are forwarded to an originating said server over the switched telephone network, are thereafter packetized and sent to a destination said server over said digital data network, and forwarded to the recipient over the switched telephone network from said destination server.

12. The network of claim 11, wherein each of said servers further comprises group message means for enabling a voice message to be sent from a caller to a plurality of recipients, and group facsimile means for enabling a facsimile transmission to be sent from a caller to a plurality of recipients over the switched telephone network and the digital data network.

13. The network of claim 12, wherein each of said servers further comprises multi-party conferencing means for permitting a caller to initiate a conference call with a plurality of recipients over the switched telephone network and the digital data network.

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